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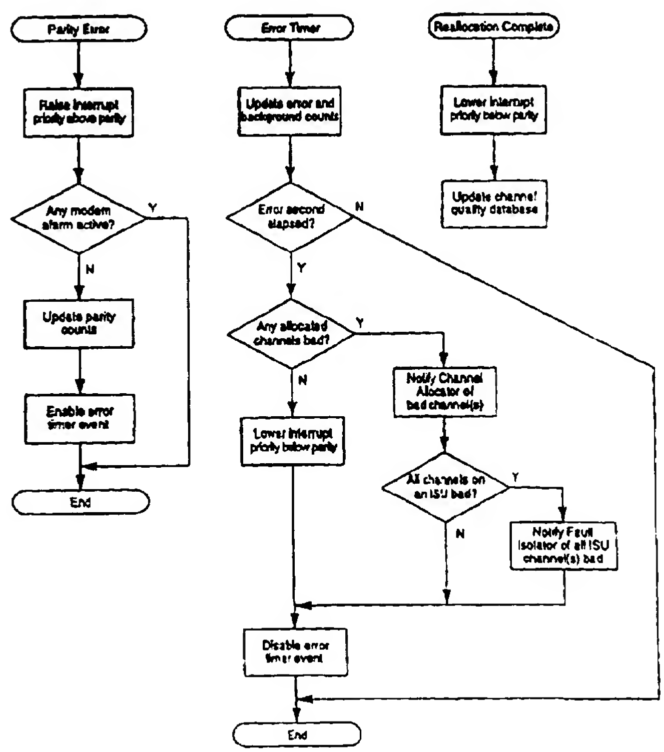
(54) Title: METHOD OF COMMUNICATION CHANNEL MONITORING USING PARITY BITS

(57) Abstract

A method for monitoring at least one telephony communication n-bit channel, wherein one of the bits is a parity bit, includes sampling the parity bit of the n-bit channel. A probable bit error rate is derived from the sampling of the parity bit. The probable bit error rate can be compared to a pre-determined bit error rate value to determine if the at least one telephony communication n-bit channel is corrupted. If the at least one telephony communication n-bit channel is corrupted, the at least one telephony communication n-bit channel is re-allocated to an uncorrupted and unallocated telephony communication n-bit channel. Further, at least one unallocated telephony communication channel can be periodically monitored and error data accumulated to indicate the quality thereof.



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METHOD OF COMMUNICATION CHANNEL MONITORING

Field of the Invention

The present invention relates generally to the field of communication systems. More particularly, the present invention relates to the monitoring of communication channels.

Background of the Invention

Two information services found in households and businesses today include television, or video, services and telephone services. Another information service involves digital data transfer which is most frequently accomplished using a modem connected to a telephone service. All further references to telephony herein shall include both telephone services and digital data transfer services.

Characteristics of telephony and video signals are different and therefore telephony and video networks are designed differently as well. For example, telephony information occupies a relatively narrow band when compared to the bandwidth for video signals. In addition, telephony signals are low frequency whereas NTSC standard video signals are transmitted at carrier frequencies greater than 50 MHz. Accordingly, telephone transmission networks are relatively narrow band systems which operate at audio frequencies and which typically serve the customer by twisted wire drops from a curb-side junction box. On the other hand, cable television services are broad band and incorporate various frequency carrier mixing methods to achieve signals compatible with conventional very high frequency television receivers. Cable television systems or video services are typically provided by cable television companies through a shielded cable service connection to each individual home or business.

One attempt to combine telephony and video services into a single network is described in U.S. Patent No. 4,977,593 to Balance entitled "Optical Communications Network." Balance describes a passive optical communications network with an optical source located in a central station. The optical source transmits time division multiplexed optical signals along an

optical fiber and which signals are later split by a series of splitters between several individual fibers servicing outstations. The network allows for digital speech data to be transmitted from the outstations to the central station via the same optical path. In addition, Balance indicates that additional wavelengths could be utilized to add services, such as cable television, via digital multiplex to the network.

A 1988 NCTA technical paper, entitled "Fiber Backbone: A Proposal For an Evolutionary Cable TV network Architecture," by James A. Chiddix and David M. Pangrac, describes a hybrid optical fiber/coaxial cable television (CATV) system architecture. The architecture builds upon existing coaxial CATV networks. The architecture includes the use of a direct optical fiber path from a head end to a number of feed points in an already existing CATV distribution system.

U.S. Patent No. 5,153,763 to Pidgeon, entitled "CATV Distribution Networks Using Light Wave Transmission Lines," describes a CATV network for distribution of broad band, multichannel CATV signals from a head end to a plurality of subscribers. Electrical to optical transmitters at the head end and optical to electrical receivers at a fiber node launch and receive optical signals corresponding to broad band CATV electrical signals. Distribution from the fiber node is obtained by transmitting electrical signals along coaxial cable transmission lines. The system reduces distortion of the transmitted broad band CATV signals by block conversion of all or part of the broad band of CATV signals to a frequency range which is less than an octave. Related U.S. Patent No. 5,262,883 to Pidgeon, entitled "CATV Distribution Networks Using Light Wave Transmission Lines," further describes the distortion reducing system.

Although the above-mentioned networks describe various concepts for transmitting broad band video signals over various architectures, which may include hybrid optical fiber/coax architectures, none of these references describe a cost effective, flexible, communications system for telephony communications. Several problems are inherent in such a communication system.

One such problem is the need to optimize the bandwidth used for transporting data so that the bandwidth used does not exceed the allotted bandwidth. Bandwidth requirements are particularly critical in multi-point to point communication where multiple transmitters at remote units must be accommodated such that allotted bandwidth is not exceeded.

A second problem involves power consumption of the system. The communication system should minimize the power used at the remote units for the transport of data, as the equipment utilized at the remote units for transmission and reception may be supplied by power distributed over the transmission medium of the system.

Data integrity must also be addressed. Both internal and external interference can degrade the communication. Internal interference exists between data signals being transported over the system. That is, transported data signals over a common communication link may experience interference therebetween, decreasing the integrity of the data. Ingress from external sources can also effect the integrity of data transmissions. A telephony communication network is susceptible to "noise" generated by external sources, such as HAM radio. Because such noise can be intermittent and vary in intensity, a method of transporting data over the system should correct or avoid the presence of such ingress.

These problems and others as will become apparent from the description to follow, present a need for an enhanced communication system.

Summary of the Invention

The use of channel monitoring to address some of the problems inherent in a multi-point to point communication system, in particular, with respect to ingress, is described. The monitoring method of the present invention monitors a telephony communication n-bit channel wherein one of the bits is a parity bit. The parity bit of the n-bit channel is sampled and a probable bit error rate is derived from the sampling of the parity bit.

In one embodiment, the probable bit error rate over a time period is compared to a predetermined bit error rate value representing a minimum bit

error rate to determine if the n-bit channel is corrupted. A corrupted channel can then either be reallocated or, in another embodiment, the transmission power of the channel can be increased to overcome the corruption.

In an alternate method embodiment, the method comprises the steps of sampling the parity bit of the n-bit channel over a first time period, deriving a probable bit error rate from the sampling of the parity bit over the first time period, comparing the probable bit error rate over the first time period to a pre-determined bit error rate value to determine if the n-bit channel is corrupted, and accumulating a probable bit error rate over a plurality of successive time periods if the n-bit channel is not corrupted.

In another alternate method embodiment, the method comprises the steps of sampling the parity bit of the n-bit channel and deriving a probable bit error rate from the sampling of the parity bit over a first time period. The probable bit error rate over the first time period is compared to a first predetermined bit error rate value to determine if the n-bit channel is corrupted. A probable bit error rate from the sampling of the parity bit over a second time period is derived. The second time period is longer than the first time period and runs concurrently therewith. The probable bit error rate over the second time period is compared to a second predetermined bit error rate value to determine if the n-bit channel is corrupted.

In still yet another alternate embodiment, a method for monitoring at least one unallocated telephony communication channel includes periodically monitoring the at least one unallocated telephony communication channel. Error data for the at least one unallocated telephony communication channel accumulated and the at least one unallocated telephony communication channel is allocated based on the error data.

Brief Description of the Drawings

Figure 1 shows a block diagram of a communication system in accordance with the present invention utilizing a hybrid fiber/coax distribution network;

Figure 2 is an alternate embodiment of the system of Figure 1;

Figure 3 is a detailed block diagram of a host digital terminal (HDT) with associated transmitters and receivers of the system of Figure 1;

Figure 4 is a block diagram of the associated transmitters and receivers of Figure 3;

Figure 5 is a block diagram of an optical distribution node of the system of Figure 1;

Figure 6 is a general block diagram of an integrated service unit (ISU) such as a home integrated service unit (HISU) or a multiple integrated service unit (MISU) of Figure 1;

Figures 7A, 7B, 7C show data frame structures and frame signaling utilized in the HDT of Figure 3;

Figure 8 is a general block diagram of a coax master card (CXMC) of a coax master unit (CXMU) of Figure 3;

Figure 9A shows a spectral allocation for a first transport embodiment for telephony transport in the system of Figure 1;

Figure 9B shows a mapping diagram for QAM modulation;

Figure 9C shows a mapping diagram for BPSK modulation;

Figure 9D shows a subband diagram for the spectral allocation of Figure 9A;

Figure 10 is a block diagram of a master coax card (MCC) downstream transmission architecture of the CXMU for the first transport embodiment of the system of Figure 1;

Figure 11 is a block diagram of a coax transport unit (CXTU) downstream receiver architecture of an MISU for the first transport embodiment of the system of Figure 1;

Figure 12 is a block diagram of a coax home module (CXHM) downstream receiver architecture of an HISU for the first transport embodiment of the of the system of Figure 1;

Figure 13 is a block diagram of a CXHM upstream transmission architecture associated with the CXHM downstream receiver architecture of Figure 12;

Figure 14 is a block diagram of a CXTU upstream transmission

architecture associated with the CXTU downstream receiver architecture of Figure 11;

Figure 15 is a block diagram of an MCC upstream receiver architecture associated with the MCC downstream transmission architecture of Figure 10;

Figure 16 is a flow diagram of a acquisition distributed loop routine for use with the system of Figure 1;

Figure 17 is a flow diagram of a tracking distributed loop architecture routine for use with the system of Figure 1;

Figure 18 shows a magnitude response of a polyphase filter bank of the MCC upstream receiver architecture of Figure 15;

Figure 19 is an enlarged view of part of the magnitude response of Figure 18;

Figure 20 is a block diagram of a ingress filter structure and FFT of the MCC upstream receiver architecture of Figure 15;

Figure 21 is a block diagram of a polyphase filter structure of the ingress filter structure and FFT of Figure 20;

Figure 22A is a block diagram of a carrier, amplitude, timing recovery block of the downstream receiver architectures of the first transport embodiment;

Figure 22B is a block diagram of a carrier, amplitude, timing recovery block of the MCC upstream receiver architecture of the first transport embodiment;

Figure 23 is a block diagram of internal equalizer operation for the receiver architectures of the first transport embodiment;

Figure 24 is a spectral allocation of a second transport embodiment for transport in the system of Figure 1;

Figure 25 is a block diagram of an MCC modem architecture of the CXMU for the second transport embodiment of the system of Figure 1;

Figure 26 is a block diagram of a subscriber modem architecture of the HISU for the second transport embodiment of the system of Figure 1;

Figure 27 is a block diagram of a modem of the subscriber modem architecture of Figure 26;

Figure 28 is a block diagram for channel monitoring used in the system of Figure 1;

Figures 29A, 29B, and 29C are flow diagrams for error monitor portions of channel monitor routines of Figure 28;

Figure 29D is an alternate flow diagram for the diagram of Figure 29B;

Figure 30 is a flow diagram for a background monitor portion of the channel monitor routines of Figure 28; and

Figure 31 is a flow diagram for a backup portion of the channel monitor routines of Figure 28.

Detailed Description of the Preferred Embodiment

The communication system 10, as shown in Figure 1, of the present invention is an access platform primarily designed to deliver residential and business telecommunication services over a hybrid fiber-coaxial (HFC) distribution network 11. The system 10 is a cost-effective platform for delivery of telephony and video services. Telephony services may include standard telephony, computer data and/or telemetry. In addition, the present system is a flexible platform for accommodating existing and emerging services for residential subscribers.

The hybrid fiber-coaxial distribution network 11 utilizes optical fiber feeder lines to deliver telephony and video service to a distribution node 18 (referred to hereinafter as the optical distribution node (ODN)) remotely located from a central office or a head end 32. From the ODNs 18, service is distributed to subscribers via a coaxial network. Several advantages exist by utilizing the HFC-based communication system 10. By utilizing fiber installed in the feeder, the system 10 spreads the cost of optoelectronics across hundreds of subscribers. Instead of having a separate copper loop which runs from a distribution point to each subscriber ("star" distribution approach), the system 10 implements a bused approach where a distribution coaxial leg 30 passes each home and subscribers "tap" the distribution coaxial leg 30 for service. The system 10 also allows non-video services to be modulated for

transmission using more cost-effective RF modem devices in dedicated portions of the RF spectrum. Finally, the system 10 allows video services to be carried on existing coaxial facilities with no additional subscriber equipment because the coaxial distribution links can directly drive existing cable-ready television sets.

It should be apparent to one skilled in the art that the modem transport architecture described herein and the functionality of the architecture and operations surrounding such architecture could be utilized with distribution networks other than hybrid fiber coax networks. For example, the functionality may be performed with respect to wireless systems. Therefore, the present invention contemplates use of such systems in accordance with the accompanying claims.

The system 10 includes host digital terminals 12 (HDTs) which implement all common equipment functions for telephony transport, such as network interface, synchronization, DS0 grooming, and operations, administration, maintenance and provisioning (OAM&P) interfaces, and which include the interface between the switching network and a transport system which carries information to and from customer interface equipment such as integrated service units 100 (ISUs). Integrated services units (ISUs) 100, such as home integrated service units (HISUs) 68 or multiple user integrated service units (MISUs) 66, which may include a business integrated service unit as opposed to a multiple dwelling integrated service unit, implement all customer interface functions and interface to the transport system which carries information to and from the switched network. In the present system, the HDT 12 is normally located in a central office and the ISUs 100 are remotely located in the field and distributed in various locations. The HDT 12 and ISUs 100 are connected via the hybrid fiber-coax distribution network 11 in a multi-point to point configuration. In the present system, the modem functionality required to transport information over the HFC distribution network 11 is performed by interface equipment in both the HDT 12 and the ISUs 100. Such modem functionality is performed utilizing orthogonal frequency division multiplexing.

The communication system shall now be generally described with reference to Figures 1, 3 and 6. The primary components of system 10 are host digital terminals (HDTs) 12, video host distribution terminal (VHDT) 34, telephony downstream transmitter 14, telephony upstream receiver 16, the hybrid fiber coax (HFC) distribution network 11 including optical distribution node 18, and integrated service units 66, 68 (shown generally as ISU 100 in Figure 6) associated with remote units 46. The HDT 12 provides telephony interface between the switching network (noted generally by trunk line 20) and the modem interface to the HFC distribution network for transport of telephony information. The telephony downstream transmitter 14 performs electrical to optical conversion of coaxial RF downstream telephony information outputs 22 of an HDT 12, shown in Figure 3, and transmits onto redundant downstream optical feeder lines 24. The telephony upstream receiver 16 performs optical to electrical conversion of optical signals on redundant upstream optical feeder lines 26 and applies electrical signals on coaxial RF upstream telephony information inputs 28 of HDT 12. The optical distribution node (ODN) 18 provides interface between the optical feeder lines 24 and 26 and coaxial distribution legs 30. The ODN 18 combines downstream video and telephony onto coaxial distribution legs 30. The integrated services units provide modem interface to the coaxial distribution network and service interface to customers.

The HDT 12 and ISUs 100 implement the telephony transport system modulator-demodulator (modem) functionality. The HDT 12 includes at least one RF MCC modem 82, shown in Figure 3 and each ISU 100 includes an RF ISU modem 101, shown in Figure 6. The MCC modems 82 and ISU modems 101 use a multi-carrier RF transmission technique to transport telephony information, such as DS0+ channels, between the HDT 12 and ISUs 100. This multi-carrier technique is based on orthogonal frequency division multiplexing (OFDM) where a bandwidth of the system is divided up into multiple carriers, each of which may represent an information channel. Multi-carrier modulation can be viewed as a technique which takes time-division multiplexed information data and transforms it to frequency-division

multiplexed data. The generation and modulation of data on multiple carriers is accomplished digitally, using an orthogonal transformation on each data channel. The receiver performs the inverse transformation on segments of the sampled waveform to demodulate the data. The multiple carriers overlap spectrally. However, as a consequence of the orthogonality of the transformation, the data in each carrier can be demodulated with negligible interference from the other carriers, thus reducing interference between data signals transported. Multi-carrier transmission obtains efficient utilization of the transmission bandwidth, particularly necessary in the upstream communication of a multi-point to point system. Multi-carrier modulation also provides an efficient means to access multiple multiplexed data streams and allows any portion of the band to be accessed to extract such multiplexed information, provides superior noise immunity to impulse noise as a consequence of having relatively long symbol times, and also provides an effective means for eliminating narrowband interference by identifying carriers which are degraded and inhibiting the use of these carriers for data transmission (such channel monitoring and protection is described in detail below). Essentially, the telephony transport system can disable use of carriers which have interference and poor performance and only use carriers which meet transmission quality targets.

Further, the ODNs 18 combine downstream video with the telephony information for transmission onto coaxial distribution legs 30. The video information from existing video services, generally shown by trunk line 20, is received by and processed by head end 32. Head end 32 or the central office, includes a video host distribution terminal 34 (VHDT) for video data interface. The VHDT 34 has optical transmitters associated therewith for communicating the video information to the remote units 46 via the ODNs 18 of the distribution network 11.

The telephony transmitter 14 of the HDTs 12, shown in Figure 3 and 4, includes two transmitters for downstream telephony transmission to protect the telephony data transmitted. These transmitters are conventional and relatively inexpensive narrow band laser transmitters. One transmitter is in

standby if the other is functioning properly. Upon detection of a fault in the operating transmitter, the transmission is switched to the standby transmitter. In contrast, the transmitter of the VHDT 34 is relatively expensive as compared to the transmitters of HDT 12 as it is a broad band analog DFB laser transmitter. Therefore, protection of the video information, a non-essential service unlike telephony data, is left unprotected. By splitting the telephony data transmission from the video data transmission, protection for the telephony data alone can be achieved. If the video data information and the telephony data were transmitted over one optical fiber line by an expensive broad band analog laser, economies may dictate that protection for telephony services may not be possible. Therefore, separation of such transmission is of importance.

Further with reference to Figure 1, the video information is optically transmitted downstream via optical fiber line 40 to splitter 38 which splits the optical video signals for transmission on a plurality of optical fiber lines 42 to a plurality of optical distribution nodes 18. The telephony transmitter 14 associated with the HDT 12 transmits optical telephony signals via optical fiber feeder line 42 to the optical distribution nodes 18. The optical distribution nodes 18 convert the optical video signals and optical telephony signals for transmission as electrical outputs via the coaxial distribution portion of the hybrid fiber coax (HFC) distribution network 11 to a plurality of remote units 46. The electrical downstream video and telephony signals are distributed to ISUs via a plurality of coaxial legs 30 and coaxial taps 44 of the coaxial distribution portion of the HFC network 11.

The remote units 46 have associated therewith an ISU 100, shown generally in Figure 6, that includes means for transmitting upstream electrical data signals including telephony information, such as from telephones and data terminals, and in addition may include means for transmitting set top box information from set top boxes 45 as described further below. The upstream electrical data signals are provided by a plurality of ISUs 100 to an optical distribution node 18 connected thereto via the coaxial portion of the HFC distribution network 11. The optical distribution node 18 converts the

upstream electrical data signals to an upstream optical data signal for transmission over an optical fiber feeder line 26 to the head end 32.

Figure 2 generally shows an alternate embodiment for providing transmission of optical video and optical telephony signals to the optical distribution nodes 18 from head end 32, the HDT 12 and VHDT 34 in this embodiment utilize the same optical transmitter and the same optical fiber feeder line 36. The signals from HDT 12 and VHDT 34 are combined and transmitted optically from headend 32 to splitter 38. The combined signal is then split by splitter 38 and four split signals are provided to the optical distribution nodes 18 for distribution to the remote units by the coaxial distribution legs 30 and coaxial taps 44. Return optical telephony signals from the ODNs 18 would be combined at splitter 38 for provision to the headend. However, as described above, the optical transmitter utilized would be relatively expensive due to its broad band capabilities, lessening the probabilities of being able to afford protection for essential telephony services.

As one skilled in the art will recognize, the fiber feeder lines 24, 26, as shown in Figure 1, may include four fibers, two for transmission downstream from downstream telephony transmitter 14 and two for transmission upstream to upstream telephony receiver 16. With the use of directional couplers, the number of such fibers may be cut in half. In addition, the number of protection transmitters and fibers utilized may vary as known to one skilled in the art and any listed number is not limiting to the present invention as described in the accompanying claims.

The present invention shall now be described in further detail. The first part of the description shall primarily deal with video transport. The remainder of the description shall primarily be with regard to telephony transport.

VIDEO TRANSPORT

The communication system 10 includes the head end 32 which receives video and telephony information from video and telephony service providers via trunk line 20. Head end 32 includes a plurality of HDTs 12 and a VHDT

34. The HDT 12 includes a network interface for communicating telephony information, such as T1, ISDN, or other data services information, to and from telephony service providers, such communication also shown generally by trunk line 20. The VHDT 34 includes a video network interface for communicating video information, such as cable TV video information and interactive data of subscribers to and from video service providers, such communication also shown generally by trunk line 20.

The VHDT 34 transmits downstream optical signals to a splitter 38 via video optical fiber feeder line 40. The passive optical splitter 38 effectively makes four copies of the downstream high bandwidth optical video signals. The duplicated downstream optical video signals are distributed to the correspondingly connected optical distribution nodes 18. One skilled in the art will readily recognize that although four copies of the downstream video signals are created, any number of copies may be made by an appropriate splitter and that the present invention is not limited to any specific number.

The splitter is a passive means for splitting broad band optical signals without the need to employ expensive broad band optical to electrical conversion hardware. Optical signal splitters are commonly known to one skilled in the art and available from numerous fiber optic component manufacturers such as Gould, Inc. In the alternative, active splitters may also be utilized. In addition, a cascaded chain of passive or active splitters would further multiply the number of duplicated optical signals for application to an additional number of optical distribution nodes and therefore increase further the remote units serviceable by a single head end. Such alternatives are contemplated in accordance with the present invention as described by the accompanying claims.

The VHDT 34 can be located in a central office, cable TV head end, or a remote site and broadcast up to about 112 NTSC channels. The VHDT 34 includes a transmission system like that of a LiteAMp™ system available from American Lightwave Systems, Inc., currently a subsidiary of the assignee hereof. Video signals are transmitted optically by amplitude modulation of a 1300 nanometer laser source at the same frequency at which the signals are

received (i.e. the optical transmission is a terahertz optical carrier which is modulated with the RF video signals). The downstream video transmission bandwidth is about 54-725 MHz. One advantage in using the same frequency for optical transmission of the video signal as the frequency of the video signals when received is to provide high bandwidth transmission with reduced conversion expense. This same-frequency transmission approach means that the modulation downstream requires optical to electrical conversion or proportional conversion with a photodiode and perhaps amplification, but no frequency conversion. In addition, there is no sample data bandwidth reduction and little loss of resolution.

An optical distribution node 18, shown in further detail in Figure 5, receives the split downstream optical video signal from the splitter 38 on optical fiber feeder line 42. The downstream optical video signal is applied to a downstream video receiver 400 of the optical distribution node 18. The optical video receiver 400 utilized is like that available in the Lite AMp™ product line available from American Lightwave Systems, Inc. The converted signal from video receiver 400, proportionally converted utilizing photodiodes, is applied to bridger amplifier 403 along with converted telephony signals from downstream telephony receiver 402. The bridger amplifier 403 simultaneously applies four downstream electrical telephony and video signals to diplex filters 406 which allow for full duplex operation by separating the transmit and receive functions when signals of two different frequency bandwidths are utilized for upstream and downstream transmission. There is no frequency conversion performed at the ODN 18 with respect to the video or the downstream telephony signals as the signals are passed through the ODNs to the remote units via the coaxial portion of the HFC distribution network 11 in the same frequency bandwidth as they are received at the ODNs 18.

After the ODN 18 has received the downstream optical video signals and such signals are converted to downstream electrical video signals, the four outputs of the ODN 18 are applied to four coaxial legs 30 of the coaxial portion of the HFC distribution network 11 for transmission of the

downstream electrical video signals to the remote units 46. Such transmission for the electrical video signals occurs in about the 54-725 MHz bandwidth. Each ODN 18 provides for the transmission on a plurality of coaxial legs 30 and any number of outputs is contemplated in accordance with the present invention as described in the accompanying claims.

As shown in Figure 1, each coaxial cable leg 30 can provide a significant number of remote units 46 with downstream electrical video and telephony signals through a plurality of coaxial taps 44. Coaxial taps are commonly known to one skilled in the art and act as passive bidirectional pickoffs of electrical signals. Each coaxial cable leg 30 may have a number of coaxial taps 44 connected in series. In addition, the coaxial portion of the HFC distribution network 11 may use any number of amplifiers to extend the distance data can be sent over the coaxial portion of such distribution network 11.

Downstream video signals are provided from the coaxial taps 44 to the remote units 46. The video signal from the coaxial tap 44 is provided to an HISU 68 which is generally shown by the block diagram of ISU 100 in Figure 6. The ISU 100 is provided with the downstream electrical video and telephony signal from tap 44 and it is applied to diplex filter 104. The downstream electrical video and telephony signal is passed through the diplex filter 104 to both an ingress filter 105 and ISU modem 101. The downstream video signal is passed by the ingress filter 105 to video equipment via an optional set top box 45. The downstream electrical telephony signal applied from the diplex filter 104 to the ISU modem 101 is processed as described in further detail below.

Ingress filter 105 provides the remote unit 46 with protection against interference of signals applied to the video equipment as opposed to those provided to other user equipment such as telephones or computer terminals. Ingress filter 105 passes the video signals; however, it blocks those frequencies not utilized by the video equipment. By blocking those frequencies not used by the video equipment, stray signals are eliminated that may interfere with the other services by the network to at least the same

remote unit.

The set top box 45 is an optional element at the remote unit 46. Interactive video data from set top box 45 would be transmitted by an additional separate RF modem provided by the video service provider at a relatively low frequency in the bandwidth of about 5 to 40 MHz. Such frequency must not be one used for the transport of upstream and downstream telephony data and downstream video.

For an MISU 66, a separate coaxial line from coaxial tap 44 is utilized to provide transmission of video signals from the coaxial tap 44 to the set top box 45 and thus for providing downstream video signals to video equipment 47. The ingress filter 105 as shown in Figure 6 is not a part of the MISU 66 as indicated by its dashed representation.

Alternative embodiments of the VHDT 34 may employ other modulation and mixing schemes or techniques to shift the video signals in frequency, and other encoding methods to transmit the information in a coded format. Such techniques and schemes for transmitting analog video data, in addition to those transmitting digital video data, are known to one skilled in the art and are contemplated in accordance with the spirit and scope of the present invention as described in the accompanying claims.

TELEPHONY TRANSPORT

With reference to Figure 3, telephony information and ISU operations and control data (hereinafter referred to as control data) modulated on carriers by MCC modem 82 is transmitted between the HDT 12 and the telephony downstream transmitter 14 via coaxial lines 22. Telephony information and control data modulated on carriers by ISUs 100 is received at telephony upstream receiver 16 and communicated to the MCC modem 82 via coaxial cable lines 28. The telephony downstream transmitter 14 and the telephony upstream receiver 16 transmit and receive, respectively, telephony information and control data via optical fiber feeder lines 24 and 26 to and from a corresponding optical distribution node 18. The control data may include all operations, administration, maintenance & provisioning (OAM&P) for

providing the telephony services of the system 11 and any other control data necessary for providing transport of telephony information between the HDT 12 and the ISUs 100.

A block diagram of the HDT 12 is shown in Figure 3. The HDT 12 includes the following modules: Eight DS1 Units (DS1U) (seven quad-DS1 units 48 plus one protection unit 50), one protection switch & test conversion unit 52 (PSTU), two clock & time slot interchange units 54 (CTSUs) (one active and one standby/protection unit), six coax master units 56 (CXMUs) (three active and three standby/protection units), two shelf control units 58 (SCNUs) (one active and one standby/protection unit), and two power supply units 60 (PWRUs) (two load-sharing units which provide the appropriate HDT voltages from a central office supply).

The HDT 12 comprises all the common equipment functions of the telephony transport of the communication system 10. The HDT 12 is normally located in a central office and directly interfaces to a local digital switch or digital network element equipment. The HDT provides the network interface 62 for all telephony information. Each HDT accommodates from 2 to 28 DSX-1 inputs at the network interface 62, representing a maximum of 672 DS0 channels.

The HDT 12 also provides all synchronization for telephony transport in the system 11. The HDT 12 may operate in any one of three synchronization modes: external timing, line timing or internal timing. External timing refers to synchronization to a building integrated timing supply reference which is sourced from a central office in which the HDT 12 is located. Line timing is synchronized to the recovered clock from a DSX-1 signal normally derived from the local digital switch. Internal timing is a free-running or hold-over operation where the HDT maintains its own synchronization in the absence of any valid reference inputs.

The HDT 12 also provides quarter-DS0 grooming capabilities and implements a 4096 x 4096 full-access, non-blocking quarter-DS0 (16 kbps) cross-connect capability. This allows DS0s and quarter-DS0s (ISDN "D" channels) to be routed from any timeslot at the DSX-1 network interface 62 to

any customer serviced by any ISU 100.

The HDT 12 further provides the RF modem functionality required for telephony transport over the HFC distribution network 11 including the MCC modem 82. The HDT 12 accommodates up to three active CXMUs 56 for providing the modem interface to the HFC distribution network 11 and also provides one-for-one protection for each active CXMU 56.

The HDT 12 coordinates the telephony transport system including control and communication of many ISUs of the multi-point to point communication system 11. Each HDT 12 module performs a function. The DS1U module 48 provides the interface to the digital network and DSX-1 termination. The PSTU 52 provides DS1U equipment protection by switching the protection DS1U 50 for a failed DS1U module 48. The CTSU 54 provides the quarter-DS0 timeslot grooming capability and all system synchronization functions. The CTSU 54 also coordinates all call processing in the system. The CXMU 56, described in further detail below, provides the modem functionality and interface for the OFDM telephony transport over the HFC distribution network 11 and the SCNU 58 supervises the operation of the entire communication system providing all OAM&P functions for telephony transport. Most processing of requests for provisioning is performed by the SCNU 58.

Downstream Telephony Transmitter

The downstream telephony transmitter 14, shown in Figure 4, takes the coaxial RF outputs 22 from the active CXMUs 56 of the HDT 12 which carry telephony information and control data and combines the outputs 22 into a downstream telephony transmission signal. The electrical-to-optical conversion logic required for the optical transmission is implemented in a stand-alone downstream telephony transmitter 14 rather than in the HDT 12 to provide a more cost effective transport solution. By placing this function in a separate component, the expense of this function does not need to be replicated in each CXMU 56 of the HDT 12. This reduces the cost of the CXMU 56 function and allows the CXMU 56 to transmit and receive over

coax instead of fiber. The downstream telephony transmitter 14 also provides for transmission on redundant downstream fiber feeder lines 24 to an ODN 18.

The downstream telephony transmitter 14 is co-located with the HDT 12 preferably within a distance of 100 feet or less. The downstream telephony transmitter 14 receives the coaxial RF outputs from the active CXMUs 56, each within a 6 MHz frequency band, and combines them at combiner 25 into a single RF signal. Each 6 MHz frequency band is separated by a guard band as is known to one skilled in the art. Downstream telephony information is then transmitted in about the 725-800 MHz frequency band. The telephony transmitter 14 passes the combined signal through a 1-to-2 splitter (not shown), thereby producing redundant downstream electrical signals. The two redundant signals are each delivered to redundant laser transmitters 501 for electrical-to-optical conversion and the redundant signals modulate an optical output such that the output of the downstream telephony transmitter 14 is on two optical feeder lines 24, each having an identical signal modulated thereon. This provides protection for the downstream telephony portion of the present system. Both Fabry-Perot lasers in the telephony transmitter 14 are active at all times. All protection functions are provided at the receive end of the optical transmission (located at the ODN 18) where one of two receivers is selected as "active;" therefore, the telephony transmitter 14 requires no protection switching capabilities.

Upstream Telephony Receiver

The upstream telephony receiver 16 performs the optical-to-electrical conversion on the upstream optical telephony signals on the upstream optical feeder lines 26 from the ODN 18. The upstream telephony receiver 16 is normally co-located in the central office with the HDT 12, and provides an electrical coaxial output to the HDT 12, and a coaxial output 23 to be provided to a video set-top controller (not shown). Upstream telephony information is routed via coax lines 28 from the upstream telephony receiver 16 to active CXMUs 56 of the HDT 12. The coaxial link 28 between the

HDT 12 and the upstream telephony receiver 16 is preferably limited to a distance of 100 feet or less and is an intra-office link. Video set-top controller information, as described in the Video Transport section hereof, is located in a bandwidth of the RF spectrum of 5-40 MHz which is not utilized for upstream telephony transport such that it is transmitted along with the upstream telephony information.

The upstream telephony receiver 16 has dual receivers 502 for the dual upstream optical fiber feeders lines 26. These feeder lines 26 carry redundant signals from the ODN 18 which contain both telephony information and control data and also video set-top box information. The upstream telephony receiver 16 performs automatic protection switching on the upstream feeder lines 26 from the ODN. The receiver 502 selected as "active" by protection logic is split to feed the coaxial outputs 28 which drive the HDT 12 and output 23 is provided to the set-top controller (not shown).

Optical Distribution Node

Referring to Figure 5, the ODN 18 provides the interface between the optical feeder lines 24 and 26 from the HDT 12 and the coaxial portion of the HFC distribution network 11 to the remote units 46. As such, the ODN 18 is essentially an optical-to-electrical and electrical-to-optical converter. The maximum distance over coax of any ISU 100 from an ODN 18 is preferably about 6 km and the maximum length of the combined optical feeder line/coaxial drop is preferably about 20 km. The optical feeder line side of the ODN 18 terminates six fibers although such number may vary. They include: a downstream video feeder line 42 (single fiber from video splitter 38), a downstream telephony feeder line 24 (from downstream telephony transmitter 14), a downstream telephony protection feeder line 24 (from downstream telephony transmitter 14), an upstream telephony feeder line 26 (to upstream telephony receiver 16), an upstream protection feeder line 26 (to upstream telephony receiver 16), and a spare fiber (not shown). The ODN 18 provides protection switching functionality on the receive optical feeder lines 24 from the downstream telephony transmitter. The ODN provides redundant

transmission on the upstream optical feeder lines 26 to the upstream telephony receiver. Protection on the upstream optical feeder lines is controlled at the upstream telephony receiver 16. On the coaxial distribution side of ODN 18, the ODN 18 terminates up to four coaxial legs 30.

In the downstream direction, the ODN 18 includes downstream telephony receiver 402 for converting the optical downstream telephony signal into an electrical signal and a bridger amplifier 403 that combines it with the converted downstream video signal from downstream video receiver 400 terminated at the ODN 18 from the VHDT 34. This combined wide-band electrical telephony/video signal is then transported in the spectrum allocated for downstream transmission, for example, the 725-800 MHz band, on each of the four coaxial legs of the coaxial portion of the HFC distribution network 11. As such, this electrical telephony and video signal is carried over the coaxial legs 30 to the ISUs 100; the bridger amplifier 403 simultaneously applying four downstream electrical telephony and video signals to diplex filters 406. The diplex filters 406 allow for full duplex operation by separating the transmit and receive functions when signals at two different frequency bandwidths are utilized for upstream and downstream transmission. There is no frequency conversion available at the ODN 18 for downstream transport as the telephony and video signals are passed through the ODN 18 to the remote units 46 via the coaxial portion of HFC distribution network 11 in the same frequency bandwidth as they are received at the ODN 18. As shown in Figure 1, each coaxial leg 30 can provide a significant number of remote units 46 with downstream electrical video and telephony signals through a plurality of coaxial taps 44. Coaxial taps 44 commonly known to one skilled in the art act as passive bidirectional pickoffs of electrical signals. Each coaxial leg 30 may have a number of coaxial taps connected in a series. In addition, the coaxial portion of the HFC distribution network 11 may use any number of amplifiers to extend the distance data can be sent over the coaxial portions of the system 10. The downstream electrical video and telephony signals are then provided to an ISU 100 (Figure 6), which, more specifically, may be an HISU 68 or an MISU 66 as shown in Figure 1.

In the upstream direction, telephony and set top box information is received by the ODN 18 at diplex filters 406 over the four coaxial legs 30 in the RF spectrum region from 5 to 40 MHz. The ODN 18 may include optional frequency shifters 64 equipped on up to three of four coaxial legs 30. These frequency shifters 64, if utilized, mix the upstream spectrum on a coaxial leg to a higher frequency prior to combining with the other three coaxial legs. Frequency shifters 64 are designed to shift the upstream spectrum in multiples of 50 MHz. For example, the frequency shifters 64 may be provisioned to mix the upstream information in the 5-40 MHz portion of the RF spectrum to any of the following ranges: 50 to 100 MHz, 100 to 150 MHz, or 150 to 200 MHz. This allows any coaxial leg 30 to use the same portion of the upstream RF spectrum as another leg without any spectrum contention when the upstream information is combined at the ODN 18. Provisioning of frequency shifters is optional on a coaxial leg 30. The ODN 18 includes combiner 408 which combines the electrical upstream telephony and set top box information from all the coaxial legs 30 (which may or may not be frequency shifted) to form one composite upstream signal having all upstream information present on each of the four coaxial legs 30. The composite electrical upstream signal is passively 1:2 split and each signal feeds an upstream Fabry-Perot laser transmitter which drives a corresponding upstream fiber feeder line 26 for transmission to the upstream telephony receiver 16.

If the upstream telephony and set top box signals are upshifted at the ODN 18, the upstream telephony receiver 16 includes frequency shifters 31 to downshift the signals according to the upshifting done at the ODN 18. A combiner 33 then combines the downshifted signals for application of a combined signal to the HDT 12. Such downshifting and combining is only utilized if the signals are upshifted at the ODN 18.

Integrated Services Unit (ISUs)

Referring to Figure 1, the ISUs 100, such as HISU 68 and MISU 66, provide the interface between the HFC distribution network 11 and the

customer services for remote units 46. Two basic types of ISUs are shown, which provide service to specific customers. Multiple user integrated service unit 66 (MISUs) may be a multiple dwelling integrated service unit or a business integrated service unit. The multiple dwelling integrated service unit may be used for mixed residential and business environments, such as multi-tenant buildings, small businesses and clusters of homes. These customers require services such as plain old telephone service (POTS), data services, DS1 services, and standard TR-57 services. Business integrated service units are designed to service business environments. They may require more services, for example, data services, ISDN, DS1 services, higher bandwidth services, such as video conferencing, etc. Home integrated services units 68 (HISUs) are used for residential environments such as single-tenant buildings and duplexes, where the intended services are POTS and basic rate integrated digital services network (ISDN). Description for ISUs shall be limited to the HISUs and MISUs for simplicity purposes as multiple dwelling and business integrated service units have similar functionality as far as the present invention is concerned.

All ISUs 100 implement RF modem functionality and can be generically shown by ISU 100 of Figure 6. ISU 100 includes ISU modem 101, coax slave controller unit (CXSU) 102, channel units 103 for providing customer service interface, and diplex filter/tap 104. In the downstream direction, the electrical downstream telephony and video signal is applied to diplex filter/tap 104 which passes telephony information to ISU modem 101 and video information to video equipment via an ingress filter 105 in the case of a HISU. When the ISU 100 is a MISU 66, the video information is rejected by the diplex filter. The ISU modem 101 demodulates the downstream telephony information utilizing a modem corresponding to the MCC modem 82 used for modulating such information on orthogonal multicarriers at HDT 12. ISU 100 demodulates downstream telephony information from a coaxial distribution leg 30 in a provisionable 6 MHz frequency band. Timing generation 107 of the ISU modem 101 provides clocking for CXSU 102 which provides processing and controls reception and

transmission by ISU modem 101. The demodulated data from ISU modem 101 is passed to the applicable channel units 103 via CXSU 102 depending upon the service provided. For example, the channel units 103 may include line cards for POTS, DS1 services, ISDN, other data services, etc. Each ISU 100 provides access to a fixed subset of all channels available in a 6 MHz frequency band corresponding to one of the CXMUs of HDT 12. This subset of channels varies depending upon the type of ISU 100. An MISU 66 may provide access to many DSO channels in a 6 MHz frequency band, while an HISU 68 may only provide access to a few DSO channels.

The channel units 103 provide telephony information and control data to the CXSU 102, which provides such data to ISU modem 101 and controls ISU modem 101 for modulation of such telephony data and control data in a provisional 6 MHz frequency band for transmission onto the coaxial distribution leg 30 connected thereto. The upstream 6 MHz frequency band provisionable for transmission by the ISU 100 to the HDT 12 corresponds to one of the downstream 6 MHz bands utilized for transmission by the CXMUs 56 of HDT 12.

The CXSU 102 which applies demodulated data from the ISU modem 101 to the applicable channel units, performs data integrity checking on the downstream 10 bit DS0+ packets received from the ISU modem 101. Each ten bit DS0+ packet as described below includes a parity or data integrity bit. The CXSU 102 will check the parity of each downstream 10 bit DS0+ channel it receives. Further, the parity of each upstream DS0+ received from the channel units 103 is calculated and a parity bit inserted as the tenth bit of the upstream DS0+ for decoding and identification by the HDT 12 of an error in the upstream data. If an error is detected by CXSU 102 when checking the parity of a downstream 10 bit DS0+ channel it receives, the parity bit of the corresponding upstream channel will be intentionally inverted to inform the HDT 12 of a parity error in the downstream direction. Therefore, the upstream parity bit is indicative of errors in the downstream DS0+ channel and the corresponding upstream DS0+ channel. An example of such a parity bit generation process is described in U.S. patent application 08/074,913

entitled "Point-to Multipoint Performance Monitoring and Failure Isolation System" assigned to the assignee hereof. This upstream parity bit is utilized in channel monitoring as described further below. As would be apparent to one skilled in the art, the parity checking and generation may be performed, at least in part, in other elements of the ISU or associated therewith such as the channel units.

Each ISU 100 recovers synchronization from downstream transmission, generates all clocks required for ISU data transport and locks these clocks to the associated HDT timing. The ISUs 100 also provide call processing functionality necessary to detect customer line seizure and line idle conditions and transmit these indications to the HDT 12. ISUs 100 terminate and receive control data from the HDT 12 and process the control data received therefrom. Included in this processing are messages to coordinate dynamic channel allocation in the communication system 10. Finally, ISUs 100 generate ISU operating voltages from a power signal received over the HFC distribution network 11 as shown by the power signal 109 taken from diplex filter/tap 104.

Data Path in HDT

The following is a detailed discussion of the data path in the host digital terminal (HDT) 12. Referring to Figure 3, the data path between the network facility at the network interface 62 and the downstream telephony transmitter 14 proceeds through the DS1U 48, CTSU 54, and CXMU 56 modules of the HDT 12, respectively, in the downstream direction. Each DS1U 48 in the HDT 12 takes four DS1s from the network and formats this information into four 24-channel, 2.56 Mbps data streams of modified DSO signals referred to as CTSU inputs 76. Each DS0 in the CTSU input has been modified by appending a ninth bit which can carry multiframe timing, signaling information and control/status messages (Figure 7A). This modified DS0 is referred to as a "DS0+." The ninth bit signal (NBS) carries a pattern which is updated each frame and repeats every 24 frames. This maps each 64 kbps DS0 from the network into a 72 kbps DS0+. Thus, the twenty-four DS0 channels available on each DS1 are formatted along with overhead

information into twenty-four DS0+ channels on each of four CTSU input streams.

The ninth bit signaling (NBS) is a mechanism developed to carry the multiframe timing, out-of-band signaling bits and miscellaneous status and control information associated with each DS0 between the DS1U and the channel units. Its main functions are to carry the signaling bits to channel units 103 and to provide a multiframe clock to the channel units 103 so that they can insert upstream bit signaling into the DS0 in the correct frame of the multiframe. Because downstream DS0s may be coming from DS1s which do not share the same multiframe phase each DS0 must carry a multiframe clock or marker which indicates the signaling frames associated with the origination DS1. The NBS provides this capability. Ninth bit signaling is transparent to the OFDM modem transport of the communication system 11.

Up to eight DS1Us 48 may be equipped in a single HDT 12; including seven active DS1Us 48 and a protection DS1U module 50. Thus, 32 CTSU inputs are connected between the DS1Us and the CTSUs 54 but a maximum of 28 can be enabled to carry traffic at any one time. The four remaining CTSU inputs are from either the protection DS1U or a failed DS1U. The PSTU includes switch control for switching the protection DS1U 50 for a failed DS1U.

Each CTSU input is capable of carrying up to 32, 10-bit channels, the first 24 channels carry DS0+s and the remaining bandwidth is unused. Each CTSU input 76 is clocked at 2.56 Mbps and is synchronized to the 8 kHz internal frame signal (Figure 7C). This corresponds to 320 bits per 125 μsec frame period. These 320 bits are framed as shown in Figure 7A. The fourteen gap bits 72 at the beginning of the frame carry only a single activity pulse in the 2nd bit position, the remaining 13 bits are not used. Of the following 288 bits, the first 216 bits normally carry twenty-four DS0+ channels where each DS0+ corresponds to a standard 64 kbps DS0 channel plus the additional 8 kbps signaling bit. Thus, each DS0+ has a bandwidth of 72 kbps (nine bits every 8 Khz frame). The remaining 72 bits are reserved for additional DS0+ payload channels. The final eighteen bits 74 of the frame

are unused gap bits.

The clock and time slot interchange unit 54 (CTSU) of the HDT 12 takes information from up to 28 active CTSU input data streams 76 and cross-connects them to up to twenty-four 32-channel, 2.56 Mbps output data streams 78 which are input to the coax master units (CXMUs) 56 of the HDT 12. The format of the data streams between the CTSU 54 and the CXMUs 56 is referred to as a CTSU output. Each CTSU output can also carry up to 32, 10-bit channels like the CTSU input. The first 28 carry traffic and the remaining bandwidth is unused. Each CTSU output is clocked at 2.56 Mbps and is synchronized to the 8 kHz internal framing signal of the HDT 12 (Figure 7C). This corresponds to 320 bits per 125 μsec frame period. The frame structure for the 320 bits are as described above for the CTSU input structure.

The HDT 12 has the capability of time and space manipulation of quarter-DS0 packets (16 kbps). This function is implemented with the time slot interchange logic that is part of CTSU 54. The CTSU implements a 4096 x 4096 quarter-DS0 cross-connect function, although not all time slots are utilized. In normal operation, the CTSU 54 combines and relocates up to 672 downstream DS0+ packets (or up to 2688 quarter-DS0 packets) arranged as 28 CTSU inputs of 24 DS0+s each, into 720 DS0+ packets (or 2880 quarter-DS0 packets) arranged as 24 CTSU outputs of 32 DS0+s each.

The system has a maximum throughput of 672 DS0+ packets at the network interface so not all of the CTSU output bandwidth is usable. If more than the 672 channels are assigned on the "CTSU output" side of the CTSU, this implies concentration is being utilized. Concentration is discussed further below.

Each CXMU 56 is connected to receive eight active CTSU outputs 78 from the active CTSU 54. The eight CTSU outputs are clocked by a 2.56 MHz system clock and each carries up to 32 DS0+s as described above. The DS0+s are further processed by the CXMU 56 and a tenth parity bit is appended to each DS0+ resulting in a 10 bit DSO+. These 10 bit packets contain the DS0, the NBS (ninth bit signal) and the parity or data integrity bit (Figure 7B). The 10 bit packets are the data transmitted on the HFC

distribution network 11 to the ISUs 100. The 10th bit or data integrity bit inserted in the downstream channels is decoded and checked at the ISU and utilized to calculate and generate a parity bit for corresponding channels in the upstream as described above. This upstream parity bit which may be representative of an error in the downstream or upstream channel is utilized to provide channel protection or monitoring as further described herein.

In the upstream direction, the reverse path through the HDT is substantially a mirror of the forward path through the HDT 12. For example, the tenth parity bit is processed at the CXMU 56 and the signal from the CXMU 56 to the CTSU 54 is in the format of Figure 7A.

The round trip delay of a DS0 is the same for every data path. The time delay over the path from the downstream CTSU output, through CXMU 56, over the HFC distribution network to the ISU 100 and then from the ISU 100, back over the HFC distribution network 11, through CXMU 56 and to CTSU 54 is controlled by upstream synchronization, as described in detail below. Generally, path delay is measured for each ISU and if it is not the correct number of frames long, the delay length is adjusted by adding delay to the path at the ISU 100.

Coax Master Unit (CXMU)

The coax master unit 56 (CXMU), shown in Figure 3, includes the coax master card logic 80 (CXMC) and the master coax card (MCC) modem 82. As previously described, up to six CXMUs may be equipped in an HDT 12. The 6 CXMUs 56 include three pairs of CXMUs 56 with each pair providing for transmit in a 6 MHz bandwidth. Each pair of CXMUs 56 includes one active CXMU and a standby CXMU. Thus, one to one protection for each CXMU is provided. As shown in Figure 3, both CXMUs of the pair are provided with upstream telephony data from the upstream telephony receiver 16 and are capable of transmitting via the coaxial line 22 to the downstream telephony transmitter 14. As such, only a control signal is required to provide for the one-to-one protection indicating which CXMU 56 of the pair is to be used for transmission or reception.

Coax Master Card Logic (CXMC)

The coax master card logic 80 (CXMC) of the CXMU 56 (Figure 8), provides the interface between the data signals of the HDT 12, in particular of the CTSU 54, and the modem interface for transport of data over the HFC distribution network 11. The CXMC 80 interfaces directly to the MCC modem 82. The CXMC 80 also implements an ISU operations channel transceiver for multi-point to point operation between the HDT 12 and all ISUs 100 serviced in the 6 MHz bandwidth in which the CXMU 56 controls transport of data within. Referring to Figure 8, the CXMC includes controller and logic 84, downstream data conversion 88, upstream data conversion 90, data integrity 92, IOC transceiver 96, and timing generator 94.

Downstream data conversion 88 performs the conversion from the nine-bit channel format from CTSU 54 (Figure 7A) to the ten-bit channel format (Figure 7B) and generates the data integrity bit in each downstream channel transported over the HFC distribution network 11. The data integrity bit represents odd parity. Downstream data conversion 80 is comprised of at least a FIFO buffer used to remove the 32 gap bits 72, 74 (Figure 7A) present in the downstream CTSU outputs and insert the tenth, data integrity bit, on each channel under control of controller and logic 84.

The upstream data conversion 90 includes at least a FIFO buffer which evaluates the tenth bit (data integrity) appended to each of the upstream channels and passes this information to the data integrity circuitry 92. The upstream data conversion 90 converts the data stream of ten-bit channels (Figure 7B) back to the nine-bit channel format (Figure 7A) for application to CTSU 54. Such conversion is performed under control of controller and logic 84.

The controller and logic 84 also manages call processing and channel allocation for the telephony transport over the HFC network 11 and maintains traffic statistics over the HFC distribution network 11 in modes where dynamic time-slot allocation is utilized, such as for providing TR-303 services, concentration services commonly known to those skilled in the art. In addition, the controller 84 maintains error statistics for the channels in the 6

MHz band in which the CXMU transports data, provides software protocol for all ISU operations channel communications, and provides control for the corresponding MCC modem 82.

The data integrity 92 circuitry processes the output of the tenth bit evaluation of each upstream channel by the upstream conversion circuit 90. In the present system, parity is only guaranteed to be valid on a provisioned channel which has a call in progress. Because initialized and activated ISU transmitters may be powered down when the ISUs are idle, the parity evaluation performed by the CXMC is not always valid. A parity error detected indicates either a transmission error in an upstream channel or a transmission error in a downstream channel corresponding to the upstream channel.

The ISU operations channel (IOC) transceiver 96 of the CXMC 80 contains transmit buffers to hold messages or control data from the controller and logic 84 and loads these IOC control messages which are a fixed total of 8 bytes in length into a 64 kbps channel to be provided to the MCC modem 82 for transport on the HFC distribution network 11. In the upstream direction, the IOC transceiver receives the 64 kbps channel via the MCC modem 82 which provides the controller and logic 84 with such messages.

The timing generator circuit 94 receives redundant system clock inputs from both the active and protection CTSUs 54 of the HDT 12. Such clocks include a 2 kHz HFC multiframe signal, which is generated by the CTSU 54 to synchronize the round trip delay on all the coaxial legs of the HFC distribution network. This signal indicates multiframe alignment on the ISU operations channel and is used to synchronize symbol timing and data reconstruction for the transport system. A 8 kHz frame signal is provided for indicating the first "gap" bit of a 2.56 MHz, 32 channel signal from the CTSU 54 to the CXMU 56. A 2.048 MHz clock is generated by the CTSU 54 to the SCNU 58 and the CXMU 56. The CXMU 56 uses this clock for ISU operations channel and modem communication between the CXMC 80 and the MCC modem 82. A 2.56 MHz bit clock is used for transfer of data signals between the DS1Us 48 and CTSUs 54 and the CTSUs 54 and CXMCs 56. A

20.48 MHz bit clock is utilized for transfer of the 10-bit data channels between the CXMC and the MCC.

Master Coax Card (MCC) Modem

The master coax card (MCC) modem 82 of the CXMU 56 interfaces on one side to the CXMC 80 and on the other side to the telephony transmitter 14 and receiver 16 for transmission on and reception from the HFC distribution network 11. The MCC modem 82 implements the modem functionality for OFDM transport of telephony data and control data. The block diagram of Figure 3 identifies the associated interconnects of the MCC modem 82 for both upstream and downstream communication. The MCC modem 82 is not an independent module in the HDT 12, as it has no interface to the HDT 12 other than through the CXMC 80 of the CXMU 56. The MCC modem 82 represents the transport system logic of the HDT 12. As such, it is responsible for implementing all requirements associated with information transport over the HFC distribution network 11. Each MCC modem 82 of the CXMUs 56 of HDT 12 is allocated a maximum bandwidth of 6 MHz in the downstream spectrum for telephony data and control data transport. The exact location of the 6 MHz band is provisionable by the CXMC 80 over the communication interface via the IOC transceiver 96 between the CXMC 80 and MCC modem 82. The downstream transmission of telephony and control data is in the RF spectrum of about 725 to 800 MHz.

Each MCC modem 82 is allocated a maximum of 6 MHz in the upstream spectrum for receipt of control data and telephony data from the ISUs within the RF spectrum of about 5 to 40 MHz. Again, the exact location of the 6 MHz band is provisionable by the CXMC 80 over the communication interface between the CXMC 80 and the MCC modem 82.

The MCC modem 82 receives 256 DS0+ channels from the CXMC 80 in the form of a 20.48 MHz signal as described previously above. The MCC modem 82 transmits this information to all the ISUs 100 using the multicarrier modulation technique based on OFDM as previously discussed herein. The MCC modem 82 also recovers 256 DS0+ multicarrier channels in the

upstream transmission over the HFC distribution network and converts this information into a 20.48 Mbps stream which is passed to CXMC 80. As described previously, the multicarrier modulation technique involves encoding the telephony and control data, such as by quadrature amplitude modulation, into symbols, and then performing an inverse fast fourier transform technique to modulate the telephony and control data on a set of orthogonal multicarriers.

Symbol alignment is a necessary requirement for the multicarrier modulation technique implemented by the MCC modem 82 and the ISU modems 101 in the ISUs 100. In the downstream direction of transmission, all information at an ISU 100 is generated by a single CXMU 56, so the symbols modulated on each multicarrier are automatically phase aligned. However, upstream symbol alignment at a receiver of the MCC modem 82 varies due to the multi-point to point nature of the HFC distribution network 11 and the unequal delay paths of the ISUs 100. In order to maximize receiver efficiency at the MCC modem 82, all upstream symbols must be aligned within a narrow phase margin. This is done by utilizing an adjustable delay parameter in each ISU 100 such that the symbol periods of all channels received upstream from the different ISUs 100 are aligned at the point they reach the HDT 12. This is part of the upstream synchronization process and shall be described further below. In addition, to maintain orthogonality of the multicarriers, the carrier frequencies used for the upstream transmission by the ISUs 100 must be frequency locked to the HDT 12.

Incoming downstream information from the CXMC 80 to the MCC modem 82 is frame aligned to the 2 kHz and 8 kHz clocks provided to the MCC modem 82. The 2 kHz multi-frame signal is used by the MCC modem 82 to convey downstream symbol timing to the ISUs as described in further detail below. This multiframe clock conveys the channel correspondence and indicates the multi-carrier frame structure so that the telephony data may be correctly reassembled at the ISU 100. Two kHz represents the greatest common factor between 10 kHz (the modem symbol rate) and 8 kHz (the data frame rate).

All ISUs 100 will use the synchronization information inserted by the associated MCC modem 82 to recover all downstream timing required by the ISUs 100. This synchronization allows the ISUs 100 to demodulate the downstream information and modulate the upstream transmission in such a way that all ISU 100 transmissions received at the HDT 12 are synchronized to the same reference. Thus, the carrier frequencies used for all ISU 100 upstream transmission will be frequency locked to the HDT 12.

The symbol alignment is performed over synchronization channels in the downstream and upstream 6 MHz bandwidths under the responsibility of the MCC modem 82, in addition to providing path delay adjustment, initialization and activation, and provisioning over such synchronization channels until initialization and activation is complete as further described herein. These parameters are then tracked by use of the IOC channels. Because of their importance in the system, the IOC channel and synchronization channels may use a different modulation scheme for transport of control data between the MCC modem 82 and ISUs 100 which is more robust or of lesser order (less bits/sec/Hz or bits/symbol) than used for transport of telephony data. For example, the telephony data may be modulated using quadrature amplitude modulation, while the IOC channel and synchronization channel may be modulated utilizing BPSK modulation techniques.

The MCC modem 82 also demodulates telephony and control data modulated on multicarriers by the ISUs 100. Such demodulation is described further below with respect to the various embodiments of the telephony transport system.

Functions with respect to the OFDM transport system for which the MCC modem 82 is responsible, include at least the following, which are further described with respect to the various embodiments in further detail. The MCC modem 82 detects a received amplitude/level of a synchronization pulse/pattern from an ISU 100 within a synchronization channel and passes an indication of this level to the CXMC 80 over the communication interface therebetween. The CXMC 80 then provides a command to the MCC modem

82 for transmission to the ISU 100 being leveled for adjustment of the amplitude level thereof. The MCC modem 82 also provides for symbol alignment of all the upstream multicarriers by correlating an upstream pattern modulated on a synchronization channel with respect to a known symbol boundary and passing a required symbol delay correction to the CXMC 80 over the communication interface therebetween. The CXMC 80 then transmits via the MCC modem 82 a message downstream to the ISU 100 to adjust the symbol delay of the ISU 100.

Likewise, with regard to synchronizing an ISU 100 for overall path delay adjustment, the MCC modem 82 correlates an upstream multiframe pattern modulated in the proper bandwidth by the ISU 100 on the IOC channel with respect to a known reference boundary, and passes a required path delay correction to the CXMC 80 over the modem interface therebetween. The CXMC 80 then transmits via the MCC modem 82 over the IOC channel a message downstream to adjust the overall path delay of an ISU 100.

Summary of Bidirectional Multi-Point to Point Telephony Transport

The following summarizes the transport of telephony and control information over the HFC distribution network 11. Each CXMU 56 of HDT 12 is provisioned with respect to its specific upstream and downstream operating frequencies. The bandwidth of both upstream and downstream transmission by the CXMU 56 are a maximum of 6 MHz, with the downstream transmission in a 6 MHz band of the RF spectrum of about 725-800 MHz.

In the downstream direction, each MCC modem 82 of the CXMU 56 provides electrical telephony and control data signals to the downstream telephony transmitter 14 via coaxial line 22 in its provisional 6 MHz bandwidth. The RF electrical telephony and control data signals from the MCC modems 82 of the HDT 12 are combined into a composite signal. The downstream telephony transmitter then passes the combined electrical signal to redundant electrical-to-optical converters for modulation onto a pair of

protected downstream optical feeder lines 24.

The downstream optical feeder lines 24 carry the telephony information and control data to an ODN 18. At the ODN 18, the optical signal is converted back to electrical and combined with the downstream video information (from the video head-end feeder line 42) into an electrical downstream RF output signal. The electrical RF output signal including the telephony information and control data is then fed to the four coaxial distribution legs 30 by ODN 18. All telephony information and control data downstream is broadcast on each coaxial leg 30 and carried over the coaxial portion of the HFC distribution network 11. The electrical downstream output RF signal is tapped from the coax and terminated on the receiver modem 101 of an ISU 100 through diplex filter 104, shown in Figure 6.

The RF electrical output signals include telephony information and control data modulated on orthogonal multicarriers by MCC modem 82 utilizing orthogonal frequency division multiplexing techniques; the telephony information and control data being mapped into symbol data and the symbols being modulated on a plurality of orthogonal carriers using fast fourier transform techniques. As the symbols are all modulated on carriers at a single point to be transmitted to multiple points in the system 11, orthogonality of the multicarriers and symbol alignment of the symbols modulated on the orthogonal multicarriers are automatically aligned for transport over the HFC distribution network 11 and the telephony information and control data is demodulated at the ISUs 100 by the modem 101.

The ISU 100 receives the RF signal tapped from the coax of the coaxial portion of the HFC network 11. The RF modem 101 of the ISU 100 demodulates the signal and passes the telephony information and control data extracted to the CXSU controller 102 for provision to channel units 103 as appropriate. The ISU 100 represents the interface where the telephony information is converted for use by a subscriber or customer.

The CXMUs 56 of the HDT 12 and the ISUs 100 implement the bidirectional multi-point to point telephony transport system of the communication system 10. The CXMUs 56 and the ISUs, therefore, carry out

the modem functionality. The transport system in accordance with the present invention may utilize three different modems to implement the modem functionality for the transport system. The first modem is the MCC modem 82 which is located in each CXMU 56 of the HDT 12. The HDT 12, for example, includes three active MCC modems 82 (Figure 3) and is capable of supporting many ISUs 100, representing a multi-point to point transport network. The MCC modem 82 coordinates telephony information transport as well as control data transport for controlling the ISUs 100 by the HDT 12. For example, the control data may include call processing messages, dynamic allocation and assignment messages, ISU synchronization control messages, ISU modem control messages, channel unit provisioning, and any other ISU operation, administration, maintenance and provisioning (OAM&P) information.

The second modem is a single family subscriber or HISU modem optimized to support a single dwelling residential unit. Therefore, it must be low in cost and low in power consumption. The third modem is the multiple subscriber or MISU modem, which is required to generally support both residential and business services.

The HISU modem and the MISU modem may take several forms. For example, the HISU modem and the MISU modem may, as described further in detail below with regard to the various embodiments of the present invention, extract only a small portion of the multicarriers transmitted from the HDT 12 or a larger portion of the multicarriers transmitted from the HDT 12. For example, the HISU may extract 20 multicarriers or 10 payload channels of telephony information transported from the HDT 12 and the MISU may extract information from 260 multicarriers or 130 payload channels transported from the HDT 12. Each of these modems may use a separate receiver portion for extracting the control data from the signal transported by the HDT 12 and an additional receiver portion of the HISU modem to extract the telephony information modulated on the multicarriers transported from the HDT 12. This shall be referred to hereinafter as an out of band ISU modem. The MCC modem 82 for use with an out of band ISU

modem may modulate control information within the orthogonal carrier waveform or on carriers somewhat offset from such orthogonal carriers. In contrast to the out of band ISU modem, the HISU and MISU modems may utilize a single receiver for the ISU modem and extract both the telephony information and control data utilizing the single receiver of the modem. This shall be referred to hereinafter as an in-band ISU modem. In such a case, the control data is modulated on carriers within the orthogonal carrier waveform but may utilize different carrier modulation techniques. For example, BPSK for modulation of control data on the carriers as opposed to modulation of telephony data on payload carriers by QAM techniques. In addition, different modulation techniques may be used for upstream or downstream transmission for both control data and telephony data. For example, downstream telephony data may be modulated on the carriers utilizing 256 QAM and upstream telephony data may be modulated on the carriers utilizing 32 QAM. Whatever modulation technique is utilized for transmission dictates what demodulation approach would be used at the receiving end of the transport system. Demodulation of the downstream telephony information and control data transported by the HDT 12 shall be explained in further detail below with reference to block diagrams of different modem embodiments.

In the upstream direction, each ISU modem 101 at an ISU 100 transmits upstream on at least one orthogonal multicarrier in a 6 MHz bandwidth in the RF spectrum of about 5 to 40 MHz; the upstream 6 MHz band corresponding to the downstream 6 MHz band in which transmissions are received. The upstream electrical telephony and control data signals are transported by the ISU modems 101 to the respectively connected optical distribution node 18 as shown in Figure 1 via the individual coaxial cable legs 30. At the ODN 18, the upstream signals from the various ISUs are combined and transmitted optically to the HDT 12 via optical feeder lines 26. As previously discussed, the upstream electrical signals from the various ISUs may, in part, be frequency shifted prior to being combined into a composite upstream optical signal. In such a case, the telephony receiver 16 would include corresponding downshifting circuitry.

Due to the multi-point to point nature of transport over the HFC distribution network 11 from multiple ISUs 100 to a single HDT 12, in order to utilize orthogonal frequency division multiplexing techniques, symbols modulated on each carrier by the ISUs 100 must be aligned within a certain phase margin. In addition, as discussed in further detail below, the round trip path delay from the network interface 62 of the HDT 12 to all ISUs 100 and back from the ISUs 100 to the network interface 62 in the communication system 10 must be equal. This is required so that signaling multiframe integrity is preserved throughout the system. In addition, a signal of proper amplitude must be received at the HDT 12 to perform any control functions with respect to the ISU 100. Likewise, with regard to OFDM transport from the ISUs 100, the ISUs 100 must be frequency locked to the HDT 12 such that the multicarriers transported over the HFC distribution network 11 are orthogonally aligned. The transport system implements a distributed loop technique for implementing this multi-point to point transport utilizing orthogonal frequency division multiplexing as further described below. When the HDT 12 receives the plurality of multicarriers which are orthogonally aligned and which have telephony and control data modulated thereon with symbols aligned, the MCC modems 82 of the CXMUs 56 demodulate the telephony information and control data from the plurality of multicarriers in their corresponding 6 MHz bandwidth and provide such telephony data to the CTSU 54 for delivery to the network interface 62 and the control data to the CXMC 80 for control of the telephony transport.

As one skilled in the art will recognize, the spectrum allocations, frequency assignments, data rates, channel numbers, types of services provided and any other parameters or characteristics of the system which may be a choice of design are to be taken as examples only. The invention as described in the accompanying claims contemplates such design choices and they therefore fall within the scope of such claims. In addition, many functions may be implemented by software or hardware and either implementation is contemplated in accordance with the scope of the claims even though reference may only be made to implementation by one or the other.

First Embodiment of Telephony Transport System

The first embodiment of the telephony transport system in accordance with the present invention shall be described with particular reference to Figures 9-23 which include block diagrams of MCC modems 82, and HISU modems and MISU modems shown generally as ISU modem 101 in Figure 6. Such modems implement the upstream and downstream modem transport functionality. Following this description is a discussion on the theory of operation utilizing such modems.

Referring to Figure 9A, the spectrum allocation for one 6 MHz band for upstream and downstream transport of telephony information and control data utilizing OFDM techniques is shown. The waveform preferably has 240 payload channels or DS0+ channels which include 480 carriers or tones for accommodating a net data rate of 19.2 Mbps, 24 IOC channels including 46 carriers or tones, and 2 synchronization channels. Each synchronization channel includes two carriers or tones and is each offset from 24 IOC channels and 240 payload channels by 10 unused carriers or tones, utilized as guard tones. The total carriers or tones is 552. The synchronization tones utilized for synchronization functions as described further below are located at the ends of the 6 MHz spectrum and the plurality of orthogonal carriers in the 6 MHz band are separated from carriers of adjacent 6 MHz bands by guard bands (516.0 kHz) at each end of the 6 MHz spectrum. The guard bands are provided at each end of the 6 MHz band to allow for filter selectivity at the transmitter and receivers of the system. The synchronization carriers are offset from the telephony data or payload carriers such that if the synchronization carrier utilized for synchronization during initialization and activation is not orthogonal with the other tones or carriers within the 6 MHz band, the synchronization signal is prevented from destroying the structure of the orthogonally aligned waveform. The synchronization tones are, therefore, outside of the main body of payload carriers of the band and interspersed IOC channels, although the synchronization channel could be considered a special IOC channel.

To minimize the power requirement of the ISUs, the amount of

bandwidth that an ISU processes is minimized. As such, the telephony payload channels and IOC channels of the 6 MHz band are interspersed in the telephony payload channels with an IOC channel located every 10 payload channels. With such a distributed technique, wherein subbands of payload channels greater than 10 include an IOC channel, the amount of bandwidth an ISU "sees" can be limited such that an IOC channel is available for the HDT 12 to communicate with the ISU 100. Such subband distribution for the spectral allocation shown in Figure 9A is shown in Figure 9D. There are 24 subbands in the 6 MHz bandwidth with each subband including 10 payload channels with an IOC channel between the 5th and 6th payload channels. A benefit of distributing the IOC channels throughout the 6 MHz band is protection from narrow band ingress. If ingress destroys an IOC channel, there are other IOC channels available and the HDT 12 can re-tune an ISU 100 to a different portion of the 6 MHz band, where an IOC channel that is not corrupted is located.

Preferably, the MISU 66 sees approximately 3 MHz of the 6 MHz bandwidth to receive up to 130 payload channels which bandwidth also includes numerous IOC channels for communication from the HDT 12 to the MISU 66. The HISU 68 sees about 100 kHz of the 6 MHz bandwidth to receive 11 channels including at least one IOC channel for communication with the HDT 12.

The primary difference between the downstream and upstream paths are the support of downstream synchronization and upstream synchronization. In the downstream direction, all ISUs lock to information from the HDT (point to multi-point). The initialization and activation of ISUs are based on signals supplied in the upstream synchronization channel. During operation, ISUs track the synchronization via the IOC channels. In the upstream, the upstream synchronization process involves the distributed (multi-point to point) control of amplitude, frequency, and timing; although frequency control can also be provided utilizing only the downstream synchronization channel as described further below. The process of upstream synchronization occurs in one of the two upstream synchronization channels, the primary or the

secondary synchronization channel.

Referring to Figure 10, the downstream transmission architecture of the MCC modem 82 is shown. Two serial data inputs, approximately 10 Mbps each, comprise the payload data from the CXMC 56 which is clocked by the 8 kHz frame clock input. The IOC control data input from the CXMC 56 is clocked by the IOC clock input, which is preferably a 2.0 kHz clock. The telephony payload data and the IOC control data enter through serial ports 132 and the data is scrambled as known to one skilled in the art by scrambler 134 to provide randomness in the waveform to be transmitted over the HFC distribution network 11. Without scrambling, very high peaks in the waveform may occur; however, if the waveform is scrambled the symbols generated by the MCC modem 82 become sufficiently random and such peaks are sufficiently limited.

The scrambled signals are applied to a symbol mapping function 136. The symbol mapping function 136 takes the input bits and maps them into a complex constellation point. For example, if the input bits are mapped into a symbol for output of a BPSK signal, every bit would be mapped to a single symbol in the constellation as in the mapping diagram for BPSK of Figure 9C. Such mapping results in inphase and quadrature values (I/Q values) for the data. BPSK is the modulation technique preferably used for the upstream and downstream IOC channels and the synchronization channels. BPSK encoding is preferred for the IOC control data so as to provide robustness in the system as previously discussed. For QPSK modulation, every two bits would map into one of four complex values that represent a constellation point. In the preferred embodiment, 32 QAM is utilized for telephony payload data, wherein every five bits of payload data is mapped into one of 32 constellation points as shown in Figure 9B. Such mapping also results in I/Q values. As such, one DS0+ signal (10 bits) is represented by two symbols and the two symbols are transmitted using two carriers. Thus, one DS0+ channel is transported over two carriers or tones of 6 MHz spectrum.

One skilled in the art will recognize that various mapping or encoding techniques may be utilized with different carriers. For example, telephony

channels carrying ISDN may be encoded using QPSK as opposed to telephony channels carrying POTS data being encoded using 32 QAM. Therefore, different telephony channels carrying different services may be modulated differently to provide for more robust telephony channels for those services that require such quality. The architecture in accordance with the present invention provides the flexibility to encode and modulate any of the channels differently from the modulation technique used for a different channel.

Each symbol that gets represented by the I/Q values is mapped into a fast fourier transform (FFT) bin of symbol buffer 138. For example, for a DS0+, running at 8 kHz frame rate, five bits are mapped into one FFT bin and five bits into another bin. Each bin or memory location of the symbol buffer 138 represents the payload data and control data in the frequency domain as I/Q values. One set of FFT bins gets mapped into the time domain through the inverse FFT 140, as is known to one skilled in the art. The inverse FFT 140 maps the complex I/Q values into time domain samples corresponding to the number of points in the FFT. Both the payload data and IOC data are mapped into the buffer 138 and transformed into time domain samples by the inverse FFT 140. The number of points in the FFT 140 may vary, but in the preferred embodiment the number of points is 256. The output of the inverse FFT 140, for a 256 point FFT, is 256 time domain samples of the waveform.

The inverse FFT 140 has separate serial outputs for inphase and quadrature (I/Q) components, FFT1 and FFT0. Digital to analog converters 142 take the inphase and quadrature components, which is a numeric representation of baseband modulated signal and convert it to a discrete waveform. The signal then passes through reconstruction filters 144 to remove harmonic content. This reconstruction is needed to avoid problems arising from multiple mixing schemes and other filtering problems. The signal is summed in a signal conversion transmitter 146 for up-converting the I/Q components utilizing a synthesized waveform that is digitally tunable with the inphase and quadrature components for mixing to the applicable transmit frequency. For example, if the synthesizer is at 600 MHz, the output

frequency will be at 600 MHz. The components are summed by the signal conversion transmitter 146 and the waveform including a plurality of orthogonal carriers is then amplified by transmitter amplifier 148 and filtered by transmitter filter 150 before being coupled onto the optical fiber by way of telephony transmitter 14. Such functions are performed under control of general purpose processor 149 and other processing circuitry of block 147 necessary to perform such modulation. The general purpose processor also receives ISU adjustment parameters from carrier, amplitude, timing recovery block 222 (Figure 15) for carrying out distributed loop symbol alignment, frequency locking, amplitude adjustment, and path delay functions as described further below.

At the downstream receiving end, either an MISU or an HISU provides for extracting telephony information and control data from the downstream transmission in one of the 6 MHz bandwidths. With respect to the MISU 66, the MISU downstream receiver architecture is shown in Figure 11. It includes a 100 MHz bandpass filter 152 to reduce the frequency band of the received 600 to 850 MHz total band broadcast downstream. The filtered signal then passes through voltage tuned filters 154 to remove out of band interference and further reduce the bandwidth. The signal is down converted to baseband frequency via quadrature and inphase down convertor 158 where the signal is mixed at complex mixers 156 utilizing synthesizer 157 which is controlled from an output of serial ports 178. The down converted I/Q components are passed through filters 159 and converted to digital format at analog to digital convertors 160. The time domain samples of the I/Q components are placed in a sample buffer 162 and a set of samples are input to down convertor compensation unit 164. The compensation unit 164 attempts to mitigate errors such as DC offsets from the mixers and differential phase delays that occur in the down conversion.

Carrier, amplitude and timing signaling are extracted from the compensated signal, by the carrier, amplitude, and timing recovery block 166 by extracting control data from the synchronization channels during initialization and activation of the ISU and the IOC channels during tracking

as further described below with reference to Figure 22A. The compensated signal in parallel form is provided to fast fourier transform (FFT) 170 to be converted into a vector of frequency domain elements which are essentially the complex constellation points with I/Q components originally created upstream at the MCC modem 82 for the DS0+ channels which the MISU sees. Due to inaccuracies in channel filtering, an equalizer 172 removes dynamic errors that occur during transmission and reception. Equalization in the upstream receiver and the downstream receiver architectures shall be explained in further detail below with reference to Figure 23. From the equalizer 172, the complex constellation points are converted to bits by symbol to bit convertor 174, descrambled at descrambler 176 which is a mirror element of scrambler 134, and the payload telephony information and IOC control data are output by the serial ports 178 to the CXSU 102 as shown in Figure 6. Block 153 includes the processing capabilities for carrying out the various functions as shown therein.

Referring to Figure 12, the HISU 68 downstream receiver architecture is shown. The primary difference between the HISU downstream receiver architecture (Figure 12) and the MISU downstream receiver architecture (Figure 11) is the amount of bandwidth being processed. The front ends of the receivers up to the FFT processing are substantially the same, except during the down conversion, the analog to digital converters 160 can be operated at a much slower rate. For instance, if the bandwidth of the signal being processed is 100 kHz, the sample rate can be approximately 200 kHz. In an MISU processing a 3 MHz signal, the sample rate is about 6 MHz. Since the HISU is limited to receiving a maximum of 10 DS0+s, the FFT 180 can be of a smaller size. A 32 point FFT 180 is preferably used in the HISU and can be implemented more efficiently, compared to a 128 or 256 point FFT utilized in the MISU. Therefore, the major difference between these architectures is that the HISU receiver architecture requires substantially less signal processing capability than the MISU receiver and as such has less power consumption. Thus, to provide a system wherein power consumption at the remote units is minimized, the smaller band of frequencies seen by the

HISU allows for such low consumption. One reason the HISU is allowed to see such a small band of carriers is that the IOC channels are interspersed throughout the 6 MHz spectrum.

Referring to Figure 13, the upstream transmission architecture for the HISU 68 is shown. The IOC control data and the telephony payload data from the CXSU 102 (Figure 6) is provided to serial ports 182 at a much slower rate in the HISU than in the MISU or HDT transmission architectures, because the HISU supports only 10 DS0+ channels. The HISU upstream transmission architecture implements three important operations. It adjusts the amplitude of the signal transmitted, the timing delay (both symbol and path delay) of the signal transmitted, and the carrier frequency of the signal transmitted. The telephony data and IOC control data enters through the serial ports 182 under control of clocking signals generated by the clock generator 173 of the HISU downstream receiver architecture, and is scrambled by scrambler 184 for the reasons stated above with regard to the MCC downstream transmission architecture. The incoming bits are mapped into symbols, or complex constellation points, including I/Q components in the frequency domain, by bits to symbol converter 186. The constellation points are then placed in symbol buffer 188. Following the buffer 188, an inverse FFT 190 is applied to the symbols to create time domain samples; 32 samples corresponding to the 32 point FFT. A delay buffer 192 is placed on the output of the inverse FFT 190 to provide multi-frame alignment at MCC modem upstream receiver architecture as a function of the upstream synchronization process controlled by the HDT 12. The delay buffer 192, therefore, provides a path delay adjustment prior to digital to analog conversion by the digital to analog converters 194 of the inphase and quadrature components of the output of the inverse FFT 190. Clock delay 196 provides a fine tune adjustment for the symbol alignment at the request of IOC control data output obtained by extracting control data from the serial stream of data prior to being scrambled. After conversion to analog components by digital to analog convertors 194, the analog components therefrom are reconstructed into a smooth analog waveform by the

reconstruction filters 198. The upstream signal is then directly up converted by direct convertor 197 to the appropriate transmit frequency under control of synthesizer block 195. Synthesizer block 195 is operated under control of commands from an IOC control channel which provides carrier frequency adjustment commands thereto as extracted in the HISU downstream receiver architecture. The up converted signal is then amplified by transmitter amplifier 200, filtered by transmitter filter 202 and transmitted upstream to be combined with other signals transmitted by other ISUs 100. The block 181 includes processing circuitry for carrying out the functions thereof.

Referring to Figure 14, the upstream transmitter architecture for the MISU 66 is shown and is substantially the same as the upstream transmitter architecture of HISU 68. However, the MISU 66 handles more channels and cannot perform the operation on a single processor as can the HISU 68. Therefore, both a processor of block 181 providing the functions of block 181 including the inverse FFT 191 and a general purpose processor 206 to support the architecture are needed to handle the increased channel capacity.

Referring to Figure 15, the MCC upstream receiver architecture of each CXMU 56 at the HDT 12 is shown. A 5 to 40 MHz band pass filter 208 filters the upstream signal which is then subjected to a direct down conversion to baseband by mixer and synthesizer circuitry 211. The outputs of the down conversion is applied to anti-alias filters 210 for conditioning thereof and the output signal is converted to digital format by analog to digital converters 212 to provide a time domain sampling of the inphase and quadrature components of the signal to narrow band ingress filter and FFT 112. The narrow band ingress filter and FFT 112, as described below, provides protection against narrow band interference that may affect the upstream transmission.

The ingress filter and FFT 112 protects ten channels at a time, therefore, if ingress affects one of the available 240 DS0+s in the 6 MHz spectrum received by MCC modem 82, a maximum of ten channels will be destroyed from the ingress. The ingress filter and FFT 112 includes a polyphase structure, as will be recognized by one skilled in the art as a common filter technique. It will be further recognized by one skilled in the art

that the number of channels protected by the polyphase filter can be varied. The output of the ingress filter and FFT 112 is coupled to an equalizer 214 which provides correction for inaccuracies that occur in the channel, such as those due to noise from reference oscillators or synthesizers. The output symbols of the equalizer 214, are applied to a symbols to bits converter 216 where the symbols are mapped into bits. The bits are provided to descramblers 218, which are a mirror of the scramblers of the ISUs 100 and the output of the descramblers are provided to serial ports 220. The output of the serial ports is broken into two payload streams and one IOC control data stream just as is provided to the MCC downstream transmitter architecture in the downstream direction. Block 217 includes the necessary processing circuitry for carrying out the functions therein.

In order to detect the downstream information, the amplitude, frequency, and timing of the arriving signal must be acquired using the downstream synchronization process. Since the downstream signal constitutes a point to multi-point node topology, the OFDM waveform arrives via a single path in an inherently synchronous manner, in contrast to the upstream signal. Acquisition of the waveform parameters is initially performed on the downstream synchronization channels in the downstream synchronization bands located at the ends of the 6 MHz spectrum. These synchronization bands include a single synchronization carrier or tone which is BPSK modulated by a 2 kHz framing clock. This tone is used to derive initial amplitude, frequency, and timing at the ISU. The synchronization carrier may be located in the center of the receive band and could be considered a special case of an IOC. After the signal is received and the receiver architecture is tuned to a typical IOC channel, the same circuitry is used to track the synchronization parameters using the IOC channel.

The process used to acquire the necessary signal parameters utilizes carrier, amplitude and timing recovery block 166 of the ISU receiver architecture, which is shown in more detail in block diagram form in Figure 22A. The carrier, amplitude and timing recovery block 166 includes a Costas loop 330 which is used to acquire the frequency lock for the received

waveform. After the signal is received from the compensation unit 164, a sample and hold 334 and analog to digital conversion 332 is applied to the signal with the resulting samples from the convertors 332 applied to the Costas loop 330. The sampling is performed under control of voltage controlled oscillator 340 as divided by divider 333 which divides by the number of points of the FFT utilized in the receiver architecture, M. The mixers 331 of the Costas loop 330 are fed by the arriving signal and the feedback path, and serve as the loop phase detectors. The output of the mixers 331 are filtered and decimated to reduce the processing requirements of subsequent hardware. Given that the received signal is band-limited, less samples are required to represent the synchronization signal. If orthogonality is not preserved in the receiver, the filter will eliminate undesired signal components from the recovery process. Under conditions of orthogonality, the LPF 337 will completely remove effects from adjacent OFDM carriers. When carrier frequency lock is achieved, the process will reveal the desired BPSK waveform in the inphase arm of the loop. The output of the decimators are fed through another mixer, then processed through the loop filter with filter function H(s) and numerically controlled oscillator (NCO), completing the feedback path to correct for frequency error. When the error is at a "small" level, the loop is locked. In order to achieve fast acquisition and minimal jitter during tracking, it will be necessary to employ dual loop bandwidths. System operation will require that frequency lock is achieved and maintained within about +/- 4% of the OFDM channel spacing (360 Hz).

The amplitude of the signal is measured at the output of the frequency recovery loop at BPSK power detector 336. The total signal power will be measured, and can be used to adjust a numerically controllable analog gain circuit (not shown). The gain circuit is intended to normalize the signal so that the analog to digital convertors are used in an optimal operating region.

Timing recovery is performed using an early-late gate type algorithm of early-late gate phase detector 338 to derive timing error, and by adjusting the sample clock or oscillator 340 in response to the error signal. The early-late gate detector results in an advance/retard command during an update

interval. This command will be applied to the sample clock or oscillator 340 through filter 341. This loop is held off until frequency lock and amplitude lock have been achieved. When the timing loop is locked, it generates a lock indicator signal. The same clocks are also used for the upstream transmission. The carrier, timing and amplitude recovery block 166 provides a reference for the clock generator 168. The clock generator 168 provides all of the clocks needed by the MISU, for example, the 8 kHz frame clock and the sample clock.

Carrier, amplitude, and timing recovery block 222 of the MCC modem upstream receiver architecture (Figure 15), is shown by the synchronization loop diagram of Figure 22B. It performs detection for upstream synchronization on signals on the upstream synchronization channel. For initialization and activation of an ISU, upstream synchronization is performed by the HDT commanding one of the ISUs via the downstream IOC control channels to send a reference signal upstream on a synchronization channel. The carrier, amplitude, and timing recovery block 222 measures the parameters of data from the ISU 100 that responds on the synchronization channel and estimates the frequency error, the amplitude error, and the timing error compared to references at the HDT 12. The output of the carrier, amplitude, and timing recovery block 222 is turned into adjustment commands by the HDT 12 and sent to the ISU being initialized and activated in the downstream direction on an IOC control channel by the MCC downstream transmitter architecture.

The purpose of the upstream synchronization process is to initialize and activate ISUs such that the waveform from distinct ISUs combine to a unified waveform at the HDT 12. The parameters that are estimated at the HDT 12 by carrier, amplitude, and timing recovery block 222 and adjusted by the ISUs are amplitude, timing, and frequency. The amplitude of an ISUs signal is normalized so that DS0+s are apportioned an equal amount of power, and achieves a desired signal to noise ratio at the HDT 12. In addition, adjacent ISUs must be received at the correct relative level or else weaker DS0+ channels will be adversely impacted by the transient behavior of the

stronger DS0+ channels. If a payload channel is transmitted adjacent to another payload channel with sufficient frequency error, orthogonality in the OFDM waveform deteriorates and error rate performance is compromised. Therefore, the frequency of the ISU must be adjusted to close tolerances. Timing of the recovered signal also impacts orthogonality. A symbol which is not aligned in time with adjacent symbols can produce transitions within the part of the symbol that is subjected to the FFT process. If the transitions of all symbols don't fall within the guard interval at the HDT, approximately +/- 16 tones (8 DS0+s) relative to the non-orthogonal channel will be unrecoverable.

During upstream synchronization, the ISUs will be commanded to send a signal, for example a square wave signal, to establish amplitude and frequency accuracy and to align symbols. The pattern signal may be any signal which allows for detection of the parameters by carrier, amplitude and timing recovery block 222 and such signal may be different for detecting different parameters. For example, the signal may be a continuous sinusoid for amplitude and frequency detection and correction and a square wave for symbol timing. The carrier, amplitude and timing recovery block 222 estimates the three distributed loop parameters. In all three loops, the resulting error signal will be converted to a command by the CXMC 80 and sent via the MCC modem 82 over an IOC channel and the CXSU will receive the command and control the adjustment made by the ISU.

As shown in Figure 22B, the upstream synchronization from the ISU is sampled and held 434 and analog to digital converted 432 under control of voltage controlled oscillator 440. Voltage controlled oscillator is a local reference oscillator which is divided by M, the points of the FFT in the receiver architecture, for control of sample and hold 434 and analog to digital convertor 432 and divided by k to apply an 8 kHz signal to phase detector 438.

Frequency error may be estimated utilizing the Costas loop 430. The Costas loop 430 attempts to establish phase lock with the locally generated frequency reference. After some period of time, loop adaptation will be

disabled and phase difference with respect to the time will be used to estimate the frequency error. The frequency error is generated by filter function H(s) 444 and provided to the CXMC 82 for processing to send a frequency adjustment command to the ISU via an IOC control channel. The frequency error is also applied to the numerically controlled oscillator (NCO) to complete the frequency loop to correct for frequency error.

The amplitude error is computed based on the magnitude of the carrier during the upstream synchronization by detecting the carrier amplitude of the inphase arm of the Costas loop 430 by power detector 436. The amplitude is compared with a desired reference value at reference comparator 443 and the error will be sent to the CXMC 82 for processing to send an amplitude adjustment command to the ISU via an IOC control channel.

When the local reference in the HDT has achieved phase lock, the BPSK signal on the synchronization channel arriving from the ISU is available for processing. The square wave is obtained on the inphase arm of the Costas loop 430 and applied to early-late gate phase detector 438 for comparison to the locally generated 8 kHz signal from divider 435. The phase detector 435 generates a phase or symbol timing error applied to loop filter 441 and output via line 439. The phase or symbol timing error is then provided to the CXMC 82 for processing to send a symbol timing adjustment command to the ISU via an IOC control channel.

The mechanisms in the ISU which adjust the parameters for upstream synchronization include implementing an amplitude change with a scalar multiplication of the time domain waveform as it is being collected from the digital processing algorithm, such as inverse FFT 190, by the digital to analog convertors 194 (Figure 13). Similarly, a complex mixing signal could be created and implemented as a complex multiply applied to the input to the digital to analog convertors 194.

Frequency accuracy of both the downstream sample clock and upstream sample clock, in the ISU, is established by phase locking an oscillator to the downstream synchronization and IOC information. Upstream transmission frequency is adjusted, for example, at synthesizer block 195 as

commanded by the HDT 12.

Symbol timing corrections are implemented as a delay function. Symbol timing alignment in the ISU upstream direction is therefore established as a delay in the sample timing accomplished by either blanking a sample interval (two of the same samples to go out simultaneously) or by putting in an extra clock edge (one sample is clocked out and lost) via clock delay 196 (Figure 13). In this manner, a delay function can be controlled without data storage overhead beyond that already required.

After the ISU is initialized and activated into the system, ready for transmission, the ISU will maintain required upstream synchronization system parameters using the carrier, amplitude, frequency recovery block 222. An unused but initialized and activated ISU will be commanded to transmit on an IOC and the block 222 will estimate the parameters therefrom as explained above.

In both the upstream transmitter architectures for the MISU 66 (Figure 13) and the HISU 68 (Figure 14), frequency offset or correction to achieve orthogonality of the carriers at HDT 12 can be determined on the ISU as opposed to the frequency offset being determined at the HDT during synchronization by carrier, amplitude and timing recovery block 222 (Figure 15) and then frequency offset adjustment commands being transmitted to the ISU for adjustment of carrier frequency via the synthesizer blocks 195 and 199 of the HISU 68 and MISU 66, respectively. Thus, frequency error would no longer be detected by carrier, amplitude and timing recovery block 22 as described above. Rather, in such a direct ISU implementation, the ISU, whether an HISU 68 or MISU 66, estimates a frequency error digitally from the downstream signal and a correction is applied to the upstream data being transmitted.

The HDT 12 derives all transmit and receive frequencies from the same fundamental oscillator. Therefore, all mixing signals are frequency locked in the HDT. Similarly, the ISU, whether an HISU 68 or MISU 66, derives all transmit and receive frequencies from the same fundamental oscillator; therefore, all the mixing signals on the ISU are also frequency

locked. There is, however, a frequency offset present in the ISU oscillators relative to the HDT oscillators. The amount of frequency error (viewed from the ISU) will be a fixed percentage of the mixing frequency. For example, if the ISU oscillator is 10PPM off in frequency from the HDT oscillators, and the downstream ISU receiver mix frequency was 100 MHz and the ISU upstream transmit mixing frequency were 10 MHz, the ISU would have to correct for 1 kHz on the downstream receiver and create a signal with a 100 Hz offset on the upstream transmitter. As such, with the ISU direct implementation, the frequency offset is estimated from the downstream signal.

The estimation is performed with digital circuitry performing numeric calculations, i.e. a processor. Samples of the synchronization channel or IOC channel are collected in hardware during operation of the system. A tracking loop drives a digital numeric oscillator which is digitally mixed against the received signal. This process derives a signal internally that is essentially locked to the HDT. The internal numerical mix accounts for the frequency offset. During the process of locking to the downstream signal in the ISU, the estimate of frequency error is derived and with the downstream frequency being known, a fractional frequency error can be computed. Based on the knowledge of the mixing frequency at the HDT that will be used to down convert the upstream receive signal, an offset to the ISU transmit frequency is computed. This frequency offset is digitally applied to the ISU transmitted signal prior to converting the signal to the analog domain, such as by convertors 194 of Figure 13. Therefore, the frequency correction can be performed directly on the ISU.

Referring to Figures 20 and 21, the narrow band ingress filter and FFT 112 of the MCC upstream receiver architecture, including a polyphase filter structure, will be described in further detail. Generally, the polyphase filter structure includes polyphase filters 122 and 124 and provides protection against ingress. The 6 MHz band of upstream OFDM carriers from the ISUs 100 is broken into subbands through the polyphase filters which provide filtering for small groups of carriers or tones, and if an ingress affects carriers within a group of carriers, only that group of carriers is affected and the other

groups of carriers are protected by such filtering characteristics.

The ingress filter structure has two parallel banks 122, 124 of polyphase filters. One bank has approximately 17 different non-overlapping bands with channel spaces between the bands. A magnitude response of a single polyphase filter bank is shown in Figure 18. The second bank is offset from the first bank by an amount so that the channels that are not filtered by the first bank are filtered by the second bank. Therefore, as shown in the closeup magnitude response of a single polyphase filter bank in Figure 19, one band of channels filtered may include those in frequency bins 38-68 with the center carriers corresponding to bins 45-61 being passed by the filter. The overlapping filter provides for filtering carriers in the spaces between the bands and the carriers not passed by the other filter bank. For example, the overlapping filter may pass 28-44. The two channel banks are offset by 16 frequency bins so that the combination of the two filter banks receives every one of the 544 channels.

Referring to Figure 20, the ingress filter structure receives the sampled waveform x(k) from the analog to digital convertors 212 and then complex mixers 118 and 120 provide the stagger for application to the polyphase filters 122, 124. The mixer 118 uses a constant value and the mixer 120 uses a value to achieve such offset. The outputs of each mixer enters one of the polyphase filters 122, 124. The output of each polyphase filter bank comprises 18 bands, each of which contain 16 usable FFT bins or each band supports sixteen carriers at the 8 kHz rate, or 8 DS0+s. One band is not utilized.

Each band output of the polyphase filters 122, 124 has 36 samples per 8 kHz frame including 4 guard samples and enters a Fast Fourier Transform (FFT) block 126, 128. The first operation performed by the FFT blocks 126, 128 is to remove the four guard samples, thereby leaving 32 time domain points. The output of the each FFT in the blocks is 32 frequency bins, 16 of which are used with the other bins providing filtering. The output of the FFTs are staggered to provide overlap. As seen in Figure 20, carriers 0 - 15 are output by FFT #1 of the top bank, carriers 16 - 31 are output by FFT #1

of the bottom bank, carriers 32 - 48 are output by FFT #2 of the top bank and so on.

The polyphase filters 122, 124 are each standard polyphase filter construction as is known to one skilled in the art and each is shown by the structure of Figure 21. The input signal is sampled at a 5.184 Mega sample per second rate, or 648 samples per frame. The input is then decimated by a factor of 18 (1 of 18 samples are kept) to give an effective sample rate of 288 kHz. This signal is subjected to the finite impulse response (FIR) filters, labeled $H_{0,0}(Z)$ through $H_{0,16}(Z)$, which include a number of taps, preferably 5 taps per filter. As one skilled in the art will recognize the number of taps can vary and is not intended to limit the scope of the invention. The outputs from the filters enter an 18 point inverse FFT 130. The output of the transform is 36 samples for an 8 kHz frame including 4 guard samples and is provided to FFT blocks 126 and 128 for processing as described above. The FFT tones are preferably spaced at 9 kHz, and the information rate is 8 kilosymbols per second with four guard samples per symbol allotted. The 17 bands from each polyphase filter are applied to the FFT blocks 126, 128 for processing and output of the 544 carriers as indicated above. One band, the 18th band, as indicated above, is not used.

The equalizer 214 (Figure 15) and 172 (Figure 11), in both upstream and downstream receiver architectures, is supplied to account for changes in group delay across the cable plant. The equalizer tracks out phase and gain or amplitude variations due to environmental changes and can therefore adapt slowly while maintaining sufficiently accurate tracking. The coefficients 360 of the equalizer 172, 214, for which the internal equalizer operation is generally shown in Figure 23, represent the inverse of the channel frequency response to the resolution of the FFT 112, 170. The downstream coefficients will be highly correlated since every channel will progress through the same signal path as opposed to the upstream coefficients which may be uncorrelated due to the variant channels that individual DS0+s may encounter in the multi-point to point topology. While the channel characteristics are diverse, the equalizer will operate the same for either upstream or downstream receiver.

The downstream equalizer will track on only the IOC channels, thus reducing the computational requirements at the ISUs and removing the requirement for a preamble in the payload channels, as described further below, since the IOC channels are always transmitted. The upstream, however, will require equalization on a per DS0+ and IOC channel basis.

The algorithm used to update the equalizer coefficients contains several local minima when operating on a 32 QAM constellation and suffers from a four-fold phase ambiguity. Furthermore, each DS0+ in the upstream can emanate from a separate ISU, and can therefore have an independent phase shift. To mitigate this problem, each communication onset will be required to post a fixed symbol preamble prior to data transmission. Note that the IOC channels are excluded from this requirement since they are not equalized and that the preamble cannot be scrambled. It is known that at the time of transmission, the HDT 12 will still have accurate frequency lock and symbol timing as established during initialization and activation of the ISU and will maintain synchronization on the continuously available downstream IOC channel.

The introduction of the preamble requires that the equalizer have knowledge of its process state. Three states are introduced which include: search, acquisition, and tracking mode. Search mode is based on the amount of power present on a channel. Transmitter algorithms will place a zero value in unused FFT bins, resulting in no power being transmitted on that particular frequency. At the receiver, the equalizer will determine that it is in search mode based on the absence of power in the FFT bin.

When transmission begins for an initialized and activated ISU, the equalizer detects the presence of signal and enters the acquisition mode. The length of the preamble may be about 15 symbols. The equalizer will vary the equalization process based on the preamble. The initial phase and amplitude correction will be large but subsequent updates of the coefficients will be less significant.

After acquisition, the equalizer will enter a tracking mode with the update rate being reduced to a minimal level. The tracking mode will

continue until a loss of power is detected on the channel for a period of time. The channel is then in the unused but initialized and activated state. The equalizer will not train or track when the receiver is being tuned and the coefficients will not be updated. The coefficients may be accessed and used such as by signal to noise detector 305 (Figure 15) for channel monitoring as discussed further below.

For the equalization process, the I/Q components are loaded into a buffer at the output of the FFT, such as FFT 112, 180. As will be apparent to one skilled in the art, the following description of the equalizer structure is with regard to the upstream receiver equalizer 214 but is equally applicable to the downstream receiver equalizer 172. The equalizer 214 extracts time domain samples from the buffer and processes one complex sample at a time. The processed information is then output therefrom. Figure 23 shows the basic structure of the equalizer algorithm less the state control algorithm which should be apparent to one skilled in the art. The primary equalization path performs a complex multiply at multiplier 370 with the value from the selected FFT bin. The output is then quantized at symbol quantize block 366 to the nearest symbol value from a storage table. The quantized value (hard decision) is passed out to be decoded into bits by symbols to bits convertor 216. The remainder of the circuitry is used to update the equalizer coefficients. An error is calculated between the quantized symbol value and the equalized sample at summer 364. This complex error is multiplied with the received sample at multiplier 363 and the result is scaled by the adaptation coefficient by multiplier 362 to form an update value. The update value is summed at summer 368 with the original coefficient to result in a new coefficient value.

Operation of First Embodiment

In the preferred embodiment, the 6 MHz frequency band for each MCC modem 82 of HDT 12 is allocated as shown in Figure 9A. Although the MCC modem 82 transmits and receives the entire 6 MHz band, the ISU modems 100 (Figure 6) are optimized for the specific application for which

they are designed and may terminate/generate fewer than the total number of carriers or tones allocated in the 6 MHz band. The upstream and downstream band allocations are preferably symmetric. The upstream 6 MHz bands from the MCC modems 82 lie in the 5-40 MHz spectrum and the downstream 6 MHz bands lie in the 725-760 MHz spectrum.

There are three regions in each 6 MHz frequency band to support specific operations, such as transport of telephony payload data, transport of ISU system operations and control data (IOC control data), and upstream and downstream synchronization. Each carrier or tone in the OFDM frequency band consists of a sinusoid which is modulated in amplitude and phase to form a complex constellation point as previously described. The fundamental symbol rate of the OFDM waveform is 8 kHz, and there are a total of 552 tones in the 6 MHz band. The following Table 1 summarizes the preferable modulation type and bandwidth allocation for the various tone classifications.

Table 1

Band Allocation	Number of Tones or Carriers	Modulation	Capacity	Bandwidth
Synch Band	24 tones(2 synch tones at each end and 10 guard tones at each end)	BPSK	n/a	216 KHz
Payload Data	480 (240 DSO + channels)	32 QAM	19.2 MBPS	4.32 MHz
IOC	48 (2 every 20 data channels or 24 IOC channels)	BPSK	384 kbps	432 kHz
Intra-band guard	Remainder on each end	n/a	n/a	1.032 MHz (516 kHz at each end)
Composite Signal	552	n/a	n/a	6.0 MHz

Guard bands are provided at each end of the spectrum to allow for selectivity filtering after transmission and prior to reception. A total of 240

telephony data channels are included throughout the band, which accommodates a net data rate of 19.2 Mbps. This capacity was designed to account for additive ingress, thereby retaining enough support to achieve concentration of users to the central office. The IOC channels are interspersed throughout the band to provide redundancy and communication support to narrowband receivers located in the HISUs. The IOC data rate is 16 kbps (two BPSK tones at the symbol rate of 8 kHz frames per second). Effectively, an IOC is provided for every 10 payload data channels. An ISU, such as an HISU, that can only see a single IOC channel would be forced to retune if the IOC channel is corrupted. However, an ISU which can see multiple IOC channels can select an alternate IOC channel in the event that the primary choice is corrupt, such as for an MISU.

The synchronization channels are duplicated at the ends of the band for redundancy, and are offset from the main body of usable carriers to guarantee that the synchronization channels do not interfere with the other used channels. The synchronization channels were previously described and will be further described below. The synchronization channels are operated at a lower power level than the telephony payload channels to also reduce the effect of any interference to such channels. This power reduction also allows for a smaller guard band to be used between the synchronization channels and the payload telephony channels.

One synchronization or redundant synchronization channels may also be implemented within the telephony channels as opposed to being offset therefrom. In order to keep them from interfering with the telephony channels, the synchronization channels may be implemented using a lower symbol rate. For example, if the telephony channels are implemented at an 8 kHz symbol rate, the synchronization channels could be implemented at a 2 kHz symbol rate and also may be at a lower power level.

The ISUs 100 are designed to receive a subband, as shown in Figure 9D, of the total aggregate 6 MHz spectrum. As an example, the HISU 68 will preferably detect only 22 of the available 552 channels. This implementation is primarily a cost/power savings technique. By reducing the number of

channels being received, the sample rate and associated processing requirements are dramatically reduced and can be achievable with common conversion parts on the market today.

A given HISU 68 is limited to receiving a maximum of 10 DS0s out of the payload data channels in the HISU receiver's frequency view. The remaining channels will be used as a guard interval. Furthermore, in order to reduce the power/cost requirements, synthesizing frequency steps will be limited to 198 kHz, limiting the HISU tuning scope to 8 channel segments. An IOC channel is provided for as shown in Figure 9D so that every HISU 68 will always see an IOC channel for control of the HISU 68 via HDT 12.

The MISU 66 is designed to receive 13 subbands, as shown in Figure 9D, or 130 of the 240 available DS0s. Again, the tuning steps will be limited to 128 kHz to realize an efficient synthesizer implementation. These are preferred values for the HISU 68 and MISU 66, and it will be noted by one skilled in the art that many of the values specified herein can be varied without changing the scope or spirit of the invention as defined by the accompanying claims.

As known to one skilled in the art, there may be need to support operation over channels in a bandwidth of less than 6 MHz. With appropriate software and hardware modifications of the system, such reconfiguration is possible as would be apparent to one skilled in the art. For example, for a 2 MHz system, in the downstream, the HDT 12 would generate the channels over a subset of the total band. The HISUs are inherently narrowband and would be able to tune into the 2 MHz band. The MISUs supporting 130 channels, would receive signals beyond the 2 MHz band. They would require reduction in filter selectivity by way of a hardware modification. An eighty (80) channel MISU would be able to operate with the constraints of the 2 MHz. system. In the upstream, the HISUs would generate signals within the 2 MHz band and the MISUs transmit section would restrict the information generated to the narrower band. At the HDT, the ingress filtering would provide sufficient selectivity against out of band signal energy. The narrowband system would require synchronization bands at the edges of the 2

MHz band.

As previously described, acquisition of signal parameters for initializing the system for detection of the downstream information is performed using the downstream synchronization channels. The ISUs use the carrier, amplitude, timing recovery block 166 to establish the downstream synchronization of frequency, amplitude and timing for such detection of downstream information. The downstream signal constitutes a point to multi-point topology and the OFDM waveform arrives at the ISUs via a single path in an inherently synchronous manner.

In the upstream direction, each ISU 100 must be initialized and activated through a process of upstream synchronization before an HDT 12 can enable the ISU 100 for transmission. The process of upstream synchronization for the ISUs is utilized so that the waveform from distinct ISUs combine to a unified waveform at the HDT. The upstream synchronization process, portions of which were previously described, involves various steps. They include: ISU transmission level adjustment, upstream multicarrier symbol alignment, carrier frequency adjustment, and round trip path delay adjustment. Such synchronization is performed after acquisition of a 6 MHz band of operation.

Generally, with respect to level adjustment, the HDT 12 calibrates the measured signal strength of the upstream transmission received from an ISU 100 and adjusts the ISU 100 transmit level so that all ISUs are within acceptable threshold. Level adjustment is performed prior to symbol alignment and path delay adjustment to maximize the accuracy of these measurements.

Generally, symbol alignment is a necessary requirement for the multicarrier modulation approach implemented by the MCC modems 82 and the ISU modems 101. In the downstream direction of transmission, all information received at the ISU 100 is generated by a single CXMU 56, so the symbols modulated on each multicarrier are automatically phase aligned. However, upstream symbol alignment at the MCC modem 82 receiver architecture varies due to the multi-point to point nature of the HFC

distribution network 11 and the unequal delay paths of the ISUs 100. In order to have maximum receiver efficiency, all upstream symbols must be aligned within a narrow phase margin. This is done by providing an adjustable delay path parameter in each ISU 100 such that the symbol periods of all channels received upstream from the different ISUs are aligned at the point they reach the HDT 12.

Generally, round trip path delay adjustment is performed such that the round trip delay from the HDT network interface 62 to all ISUs 100 and back to the network interface 62 from all the ISUs 100 in a system must be equal. This is required so that signaling multiframe integrity is preserved throughout the system. All round trip processing for the telephony transport section has a predictable delay with the exception of the physical delay associated with signal propagation across the HFC distribution network 11 itself. ISUs 100 located at close physical distance from the HDT 12 will have less round trip delay than ISUs located at the maximum distance from the HDT 12. Path delay adjustment is implemented to force the transport system of all ISUs to have equal round trip propagation delay. This also maintains DS1 multiframe alignment for DS1 channels transported through the system, maintaining in-band channel signaling or robbed-bit signaling with the same alignment for voice services associated with the same DS1.

Generally, carrier frequency adjustment must be performed such that the spacing between carrier frequencies is such as to maintain orthogonality of the carriers. If the multicarriers are not received at the MCC modem 82 in orthogonal alignment, interference between the multicarriers may occur. Such carrier frequency adjustment can be performed in a manner like that of symbol timing or amplitude adjustment or may be implemented on the ISU as described previously above.

In the initialization process, when the ISU has just been powered up, the ISU 100 has no knowledge of which downstream 6 MHz frequency band it should be receiving in which provides the need for the acquisition of 6 MHz band for operation step of the initialization process. Until an ISU 100 has successfully acquired a 6 MHz band for operation, it implements a "scanning"

approach to locate its downstream frequency band. A local processor of the CXSU controller 102 of ISU 100 starts with a default 6 MHz receive frequency band somewhere in the range from 625 to 850 MHz. The ISU 100 waits for a period of time, for example 100 milliseconds, in each 6 MHz band to look for a valid 6 MHz acquisition command which matches a unique identification number for the ISU 100; which unique identifier may take the form of or be based on a serial number of the ISU equipment. If a valid 6 MHz acquisition command is not found in that 6 MHz band, the CXSU controller 102 looks at the next 6 MHz band and the process is repeated. In this manner, as explained further below, the HDT 12 can tell the ISU 100 which 6 MHz band it should use for frequency reception and which band for frequency transmission upstream.

The process of initialization and activation of ISUs, as generally described above, and tracking or follow-up synchronization is further described below. This description is written using an MISU 66 in conjunction with a CXSU controller 103 but is equally applicable to any ISU 100 implemented with an equivalent controller logic. The coax master card logic (CXMC) 80 is instructed by the shelf controller unit (SCNU) 58 to initialize and activate a particular ISU 100. The SCNU uses an ISU designation number to address the ISU 100. The CXMC 80 correlates the ISU designation number with an equipment serial number, or unique identifier, for the equipment. No two ISU equipments shipped from the factory possess the same unique identifier. If the ISU 100 has never before been initialized and activated in the current system database, the CXMC 80 chooses a personal identification number (PIN) code for the ISU 100 being initialized and activated. This PIN code is then stored in the CXMC 80 and effectively represents the "address" for all communications with that ISU 100 which will follow. The CXMC 80 maintains a lookup table between each ISU designation number, the unique identifier for the ISU equipment, and the PIN code. Each ISU 100 associated with the CXMU 56 has a unique PIN address code assignment. One PIN address code will be reserved for a broadcast feature to all ISUs, which allows for the HDT to send messages to all

initialized and activated ISUs 100.

The CXMC 80 sends an initialization and activation enabling message to the MCC modem 82 which notifies the MCC modem 82 that the process is beginning and the associated detection functionality in the MCC modem 82 should be enabled. Such functionality is performed at least in part by carrier, amplitude, timing recovery block 222 as shown in the MCC upstream receiver architecture of Figure 15 and as previously discussed.

The CXMC 80 transmits an identification message by the MCC modem 82 over all IOC channels of the 6 MHz band in which it transmits. The message includes a PIN address code to be assigned to the ISU being initialized and activated, a command indicating that ISU initialization and activation should be enabled at the ISU 100, the unique identifier for the ISU equipment, such as the equipment serial number, and cyclical redundancy checksum (CRC). The messages are sent periodically for a certain period of time. This period of time being the maximum time which an ISU can scan all downstream 6 MHz bands, listening for a valid identification message. The periodic rate, for example 50 msec, affects how quickly the ISU learns its identity. The CXMC 80 will never attempt to synchronize more than one ISU at a time. A software timeout is implemented if an ISU does not respond after some maximum time limit is exceeded. This timeout must be beyond the maximum time limit required for an ISU to obtain synchronization functions.

During periodic transmission by CXMC 80, the ISU implements the scanning approach to locate its downstream frequency band. The local processor of the CXSU starts with a default 6 MHz receive frequency band somewhere in the range from 625 to 850 MHz. The ISU 100 selects the primary synchronization channel of the 6 MHz band and then tests for loss of synchronization after a period of time. If loss of synchronization is still present, the secondary synchronization channel is selected and tested for loss of synchronization after a period of time. If loss of synchronization is still present, then the ISU restarts selection of the synchronization channels on the next 6 MHz band. When loss of synchronization is not present on a

synchronization channel then the ISU selects the first subband including the IOC and listens for a correct identification message. If a correct identification message is found which matches its unique identifier then the PIN address code is latched into an appropriate register. If a correct identification message is not found in the first subband then a middle subband is selected, such as the 11th subband, and the ISU again listens for the correct identification message. If the message again is incorrect, then the ISU restarts on another 6 MHz band. The ISU listens for the correct identification message in a subband for a period of time equal to at least two times the CXMU transmission time, for example 100 msec when transmission time is 50 msec as described above. The initialization and activation commands are unique commands in the ISU 100, as the ISU 100 will not require a PIN address code match to respond to such commands, but only a valid unique identifier and CRC match. However, the initialization and activation command from the CXMC 80 transmitted via the MCC modem 82 will be the only command which an ISU 100 will be allowed to receive without a valid PIN address code match. If an un-initialized and un-activated ISU 100 receives an initialization and activation command from the CXMC 80 via the MCC modem 82 on an IOC channel, data which matches the unique identifier and a valid CRC, the CXSU 102 of the ISU 100 will store the PIN address code transmitted with the command and the unique identifier. From this point on, the ISU 100 will only respond to commands which address it by its correct PIN address code, or a broadcast address code; unless, of course, the ISU is re-activated again and given a new PIN address code.

After the ISU 100 has received a match to its unique identifier, the ISU will receive the upstream frequency band command with a valid PIN address code that tells the ISU 100 which 6 MHz band to use for upstream transmission and the carrier or tone designations for the upstream IOC channel to be used by the ISU 100. The CXSU controller 102 interprets the command and correctly activates the ISU modem 101 of the ISU 100 for the correct upstream frequency band to respond in. Once the ISU modem 101 has acquired the correct 6 MHz band, the CXSU controller 103 sends a message

command to the ISU modem 101 to enable upstream synchronization. Distributed loops utilizing the carrier, amplitude, and timing recovery block 222 of the MCC modem upstream receiver architecture of the HDT 12 is used to lock the various ISU parameters for upstream transmission, including amplitude, carrier frequency, symbol alignment, and path delay.

Figure 16 describes this distributed loop generally. When a new unit is hooked to a cable, the HDT 12 instructs the ISU hooked to the cable to go into an upstream synchronization mode exclusive of any other ISU 100. The HDT is then given information on the new ISU and provides downstream commands for the various parameters to the subscriber ISU unit. The ISU begins transmission in the upstream and the HDT 12 locks to the upstream signal. The HDT 12 derives an error indicator with regard to the parameter being adjusted and commands the subscriber ISU to adjust such parameter. The adjustment of error is repeated in the process until the parameter for ISU transmission is locked to the HDT 12.

More specifically, after the ISU 100 has acquired the 6 MHz band for operation, the CXSU 102 sends a message command to the ISU modem 101 and the ISU modem 101 transmits a synchronization pattern on a synchronization channel in the primary synchronization band of the spectral allocation as shown in Figure 9. The upstream synchronization channels which are offset from the payload data channels as allocated in Figure 9 include both a primary and a redundant synchronization channel such that upstream synchronization can still be accomplished if one of the synchronization channels is corrupted.

The MCC modem 82 detects a valid signal and performs an amplitude level measurement on a received signal from the ISU. The synchronization pattern indicates to the CXMC 80 that the ISU 100 has received the activation and initialization and frequency band commands and is ready to proceed with upstream synchronization. The amplitude level is compared to a desired reference level. The CXMC 80 determines whether or not the transmit level of the ISU 100 should be adjusted and the amount of such adjustment. If level adjustment is required, the CXMC 80 transmits a message on the

downstream IOC channel instructing the CXSU 102 of the ISU 100 to adjust the power level of the transmitter of the ISU modem 101. The CXMC 80 continues to check the receive power level from the ISU 100 and provides adjustment commands to the ISU 100 until the level transmitted by the ISU 100 is acceptable. The amplitude is adjusted at the ISU as previously discussed. If amplitude equilibrium is not reached within a certain number of iterations of amplitude adjustment or if a signal presence is never detected utilizing the primary synchronization channel then the same process is used on the redundant synchronization channel. If amplitude equilibrium is not reached within a certain number of iterations of amplitude adjustment or if a signal presence is never detected utilizing the primary or redundant synchronization channels then the ISU is reset.

Once transmission level adjustment of the ISU 100 is completed and has been stabilized, the CXMC 80 and MCC modem 82 perform carrier frequency locking. The MCC modem 82 detects the carrier frequency as transmitted by the ISU 100 and performs a correlation on the received transmission from the ISU to calculate a carrier frequency error correction necessary to orthogonally align the multicarriers of all the upstream transmissions from the ISUs. The MCC modem 82 returns a message to the CXMC 80 indicating the amount of carrier frequency error adjustment required to perform frequency alignment for the ISU. The CXMC 80 transmits a message on a downstream IOC channel via the MCC modem 82 instructing the CXSU 102 to adjust the transmit frequency of the ISU modem 101 and the process is repeated until the frequency has been established to within a certain tolerance for the OFDM channel spacing. Such adjustment would be made via at least the synthesizer block 195 (Figures 13 and 14). If frequency locking and adjustment is accomplished on the ISU as previously described, then this frequency adjustment method is not utilized.

To establish orthogonality, the CXMC 80 and MCC modem 82 perform symbol alignment. The MCC modem 82 detects the synchronization channel modulated at a 8 kHz frame rate transmitted by the ISU modem 101 and performs a hardware correlation on the receive signal to calculate the

delay correction necessary to symbol align the upstream ISU transmission from all the distinct ISUs 100. The MCC modem 82 returns a message to the CXMC 80 indicating the amount of delay adjustment required to symbol align the ISU 100 such that all the symbols are received at the HDT 12 simultaneously. The CXMC 80 transmits a message in a downstream IOC channel by the MCC modem 82 instructing the CXSU 103 to adjust the delay of the ISU modem 101 transmission and the process repeats until ISU symbol alignment is achieved. Such symbol alignment would be adjusted via at least the clock delay 196 (Figures 13 and 14). Numerous iterations may be necessary to reach symbol alignment equilibrium and if it is not reached within a predetermined number of iterations, then the ISU may again be reset.

Simultaneously with symbol alignment, the CXMC 80 transmits a message to the MCC modem 82 to perform path delay adjustment. The CXMC 80 sends a message on a downstream IOC channel via the MCC modem 82 instructing the CXSU controller 102 to enable the ISU modem 101 to transmit a another signal on a synchronization channel which indicates the multiframe (2 kHz) alignment of the ISU 100. The MCC modem 82 detects this multiframe alignment pattern and performs a hardware correlation on the pattern. From this correlation, the modem 82 calculates the additional symbol periods required to meet the round trip path delay of the communication system. The MCC modem 82 then returns a message to the CXMC 80 indicating the additional amount of delay which must be added to meet the overall path delay requirements and the CXMC then transmits a message on a downstream IOC channel via the MCC modem 82 instructing the CXSU controller 102 to relay a message to the ISU modem 101 containing the path delay adjustment value. Numerous iterations may be necessary to reach path delay equilibrium and if it is not reached within a predetermined number of iterations, then the ISU may again be reset. Such adjustment is made in the ISU transmitter as can be seen in the display delay buffer "n" samples 192 of the upstream transmitter architectures of Figures 13 and 14. Path delay and symbol alignment may be performed at the same time, separately or together

using the same or different signals sent on the synchronization channel.

Until the ISU is initialized and activated, the ISU 100 has no capability of transmitting telephony data information on any of the 480 tones or carriers. After such initialization and activation has been completed, the ISUs are within tolerance required for transmission within the OFDM waveform and the ISU is informed that transmission is possible and upstream synchronization is complete.

After an ISU 100 is initialized and activated for the system, follow-up synchronization or tracking may be performed periodically to keep the ISUs calibrated within the required tolerance of the OFDM transport requirements. The follow-up process is implemented to account for drift of component values with temperature. If an ISU 100 is inactive for extreme periods of time, the ISU can be tuned to the synchronization channels and requested to update upstream synchronization parameters in accordance with the upstream synchronization process described above. Alternatively, if an ISU has been used recently, the follow-up synchronization or tracking can proceed over an IOC channel. Under this scenario, as generally shown in Figure 17, the ISU 100 is requested to provide a signal over an IOC channel by the HDT 12. The HDT 12 then acquires and verifies that the signal is within the tolerance required for a channel within the OFDM waveform. If not than the ISU is requested to adjust such errored parameters. In addition, during long periods of use, ISUs can also be requested by the HDT 12 to send a signal on an IOC channel or a synchronization channel for the purpose of updating the upstream synchronization parameters.

In the downstream direction, the IOC channels transport control information to the ISUs 100. The modulation format is preferably differentially encoded BPSK, although the differential aspect of the downstream modulation is not required. In the upstream direction, the IOC channels transport control information to the HDT 12. The IOC channels are differentially BPSK modulated to mitigate the transient time associated with the equalizer when sending data in the upstream direction. Control data is slotted on a byte boundary (500 μs frame). Data from any ISU can be

transmitted on an IOC channel asynchronously; therefore, there is the potential for collisions to occur.

As there is potential for collisions, detection of collisions on the upstream IOC channels is accomplished at a data protocol level. The protocol for handling such collisions may, for example, include exponential backoff by the ISUs. As such, when the HDT 12 detects an error in transmission, a retransmission command is broadcast to all the ISUs such that the ISUs retransmit the upstream signal on the IOC channel after waiting a particular time; the wait time period being based on an exponential function.

One skilled in the art will recognize that upstream synchronization can be implemented allowing for multi-point to point transmission using only the symbol timing loop for adjustment of symbol timing by the ISUs as commanded by the HDT. The frequency loop for upstream synchronization can be eliminated with use of high quality local free running oscillators in the ISUs that are not locked to the HDT. In addition, the local oscillators at the ISUs could be locked to an outside reference. The amplitude loop is not essential to achieve symbol alignment at the HDT.

Call processing in the communication system 10 entails the manner in which a subscriber is allocated channels of the system for telephony transport from the HDT 12 to the ISUs 100. The present communication system in accordance with the present invention is capable of supporting both call processing techniques not involving concentration, for example, TR-8 services, and those involving concentration, such as TR-303 services. Concentration occurs when there are more ISU terminations requiring service than there are channels to service such ISUs. For example, there may be 1,000 customer line terminations for the system, with only 240 payload channels which can be allocated to provide service to such customers.

Where no concentration is required, such as for TR-8 operation, channels within the 6 MHz spectrum are statically allocated. Therefore, only reallocation of channels shall be discussed further below with regard to channel monitoring.

On the other hand, for dynamically allocated channels to provide

concentration, such as for providing TR-303 services, the HDT 12 supports on-demand allocation of channels for transport of telephony data over the HFC distribution network 11. Such dynamic allocation of channels is accomplished utilizing the IOC channels for communication between the HDT 12 and the ISUs 100. Channels are dynamically allocated for calls being received by a customer at an ISU 100, or for calls originated by a customer at an ISU 100. The CXMU 56 of HDT 12, as previously discussed, implements IOC channels which carry the call processing information between the HDT 12 and the ISUs 100. In particular, the following call processing messages exist on the IOC channels. They include at least a line seizure or off-hook message from the ISU to the HDT; line idle or on-hook message from the ISU to the HDT; enable and disable line idle detection messages between the HDT and the ISUs.

For a call to a subscriber on the HFC distribution network 11, the CTSU 54 sends a message to the CXMU 56 associated with the subscriber line termination and instructs the CXMU 56 to allocate a channel for transport of the call over the HFC distribution network 11. The CXMU 56 then inserts a command on the IOC channel to be received by the ISU 100 to which the call is intended; the command providing the proper information to the CXSU 102 to alert the ISU 100 as to the allocated channel.

When a call is originated by a subscriber at the ISU side, each ISU 100 is responsible for monitoring the channel units for line seizure. When line seizure is detected, the ISU 100 must communicate this change along with the PIN address code for the originating line to the CXMU 56 of the HDT 12 using the upstream IOC operation channel. Once the CXMU 56 correctly receives the line seizure message, the CXMU 56 forwards this indication to the CTSU 54 which, in turn, provides the necessary information to the switching network to set up the call. The CTSU 54 checks the availability of channels and allocates a channel for the call originated at the ISU 100. Once a channel is identified for completing the call from the ISU 100, the CXMU 56 allocates the channel over the downstream IOC channel to the ISU 100 requesting line seizure. When a subscriber returns on hook, an appropriate

line idle message is sent upstream to the HDT 12 which provides such information to the CTSU 54 such that the channel can then be allocated again to support TR-303 services.

Idle channel detection can further be accomplished in the modem utilizing another technique. After a subscriber at the ISU 100 has terminated use of a data payload channel, the MCC modem 82 can make a determination that the previously allocated channel is idle. Idle detection may be performed by utilizing the equalization process by equalizer 214 (Figure 15) which examines the results of the FFT which outputs a complex (I and Q component) symbol value. An error is calculated, as previously described herein with respect to equalization, which is used to update the equalizer coefficients. Typically, when the equalizer has acquired the signal and valid data is being detected, the error will be small. In the event that the signal is terminated, the error signal will increase, and this can be monitored by signal to noise monitor 305 to determine the termination of the payload data channel used or channel idle status. This information can then be utilized for allocating idle channels when such operation of the system supports concentration.

The equalization process can also be utilized to determine whether an unallocated or allocated channel is being corrupted by ingress as shall be explained in further detail below with respect to channel monitoring.

The telephony transport system may provide for channel protection from ingress in several manners. Narrowband ingress is a narrowband signal that is coupled into the transmission from an external source. The ingress signal which is located within the OFDM waveform can potentially take the entire band offline. An ingress signal is (most likely) not orthogonal to the OFDM carriers, and under worst case conditions can inject interference into every OFDM carrier signal at a sufficient level to corrupt almost every DS0+ to an extent that performance is degraded below a minimum bit error rate.

One method provides a digitally tunable notch filter which includes an interference sensing algorithm for identifying the ingress location on the frequency band. Once located, the filtering is updated to provide an arbitrary

filter response to notch the ingress from the OFDM waveform. The filter would not be part of the basic modem operation but requires the identification of channels that are degraded in order to "tune" them out. The amount of channels lost as a result of the filtering would be determined in response to the bit error rate characteristics in a frequency region to determine how many channels the ingress actually corrupted.

Another approach as previously discussed with respect to the ingress filter and FFT 112 of the MCC upstream receiver architecture of Figure 15 is the polyphase filter structure. The cost and power associated with the filter are absorbed at the HDT 12, while supplying sufficient ingress protection for the system. Thus, power consumption at the ISUs 100 is not increased. The preferred filter structure involves two staggered polyphase filters as previously discussed with respect to Figures 20 and 21 although use of one filter is clearly contemplated with loss of some channels. The filter/transform pair combines the filter and demodulation process into a single step. Some of the features provided by polyphase filtering include the ability to protect the receive band against narrowband ingress and allow for scalable bandwidth usage in the upstream transmission. With these approaches, if ingress renders some channels unusable, the HDT 12 can command the ISUs to transmit upstream on a different carrier frequency to avoid such ingress.

The above approaches for ingress protection, including at least the use of digital tunable notch filters and polyphase filters, are equally applicable to point to point systems utilizing multicarrier transport. For example, a single MISU transporting to a single HDT may use such techniques. In addition, uni-directional multi-point to point transport may also utilize such techniques for ingress protection.

In addition, channel monitoring and allocation or reallocation based thereon may also be used to avoid ingress. External variables can adversely affect the quality of a given channel. These variables are numerous, and can range from electro-magnet interference to a physical break in an optical fiber. A physical break in an optical fiber severs the communication link and cannot be avoided by switching channels, however, a channel which is electrically

interfered with can be avoided until the interference is gone. After the interference is gone the channel could be used again.

Referring to Figure 28, a channel monitoring method is used to detect and avoid use of corrupted channels. A channel monitor 296 is used to receive events from board support software 298 and update a channel quality table 300 in a local database. The monitor 296 also sends messages to a fault isolator 302 and to channel allocator 304 for allocation or reallocation. The basic input to the channel monitor is parity errors which are available from hardware per the upstream DS0+ channels; the DS0+ channels being 10-bit channels with one of the bits being a parity or data integrity bit inserted in the channel as previously discussed. The parity error information on a particular channel is used as raw data which is sampled and integrated over time to arrive at a quality status for that channel.

Parity errors are integrated using two time frames for each of the different service types including POTS, ISDN, DDS, and DS1, to determine channel status. The first integration routine is based on a short integration time of one second for all service types. The second routine, long integration, is service dependent, as bit error rate requirements for various services require differing integration times and monitoring periods as seen in Table 3. These two methods are described below.

Referring to Figure 29A, 29B, and 29C, the basic short integration operation is described. When a parity error of a channel is detected by the CXMU 56, a parity interrupt is disabled by setting the interrupt priority level above that of the parity interrupt (Figure 29A). If a modem alarm is received which indicates a received signal failure, parity errors will be ignored until the failure condition ends. Thus, some failure conditions will supersede parity error monitoring. Such alarm conditions may include loss of signal, modem failure, and loss of synchronization. If a modem alarm is not active, a parity count table is updated and an error timer event as shown in Figure 29B is enabled.

When the error timer event is enabled, the channel monitor 296 enters a mode wherein parity error registers of the CXMU 56 are read every 10

milliseconds and error counts are summarized after a one second monitoring period. Generally, the error counts are used to update the channel quality database and determine which (if any) channels require re-allocation. The channel quality table 300 of the database contains an ongoing record of each channel. The table organizes the history of the channels in categories such as: current ISU assigned to the channel, start of monitoring, end of monitoring, total error, errors in last day, in last week and in last 30 days, number of seconds since last error, severe errors in last day, in last week and in last 30 days, and current service type, such as ISDN, assigned to the channel.

As indicated in Figure 29A, after the parity interrupt is disabled and no active alarm exists, the parity counts are updated and the timer event is enabled. The timer event (Figure 29B), as indicated above, includes a one second loop where the errors are monitored. As shown in Figure 29B, if the one second loop has not elapsed, the error counts are continued to be updated. When the second has elapsed, the errors are summarized. If the summarized errors over the one second period exceed an allowed amount indicating that an allocated channel is corrupted or bad, as described below, channel allocator 304 is notified and ISU transmission is reallocated to a different channel. As shown in Figure 29C, when the reallocation has been completed, the interrupt priority is lowered below parity so that channel monitoring continues and the channel quality database is updated concerning the actions taken. The reallocation task may be accomplished as a separate task from the error timer task or performed in conjunction with that task. For example, the reallocator 304 may be part of channel monitor 296.

As shown in Figure 29D in an alternate embodiment of the error timer task of Figure 29B, channels can be determined to be bad before the one second has elapsed. This allows the channels that are determined to be corrupted during the initial portion of a one second interval to be quickly identified and reallocated without waiting for the entire one second to elapse.

Instead of reallocation, the power level for transmission by the ISU may be increased to overcome the ingress on the channel. However, if the power level on one channel is increased, the power level of at least one other

channel must be decreased as the overall power level must be kept substantially constant.

If all channels are determined bad, the fault isolator 302 is notified indicating the probability that a critical failure is present, such as a fiber break. If the summarized errors over the one second period do not exceed an allowed amount indicating that the allocated channel is not corrupted, the interrupt priority is lowered below parity and the error timer event is disabled. Such event is then ended and the channels once again are monitored for parity errors per Figure 29A.

Two issues presented by periodic parity monitoring as described above must be addressed in order to estimate the bit error rate corresponding to the observed count of parity errors in a monitoring period of one second to determine if a channel is corrupted. The first is the nature of parity itself. Accepted practice for data formats using block error detection assumes that an errored block represents one bit of error, even though the error actually represents a large number of data bits. Due to the nature of the data transport system, errors injected into modulated data are expected to randomize the data. This means that the average errored frame will consist of four (4) errored data bits (excluding the ninth bit). Since parity detects only odd bit errors, half of all errored frames are not detected by parity. Therefore, each parity (frame) error induced by transport interference represents an average of 8 (data) bits of error. Second, each monitoring parity error represents 80 frames of data (10 ms/125 μs). Since the parity error is latched, all errors will be detected, but multiple errors will be detected as one error.

The bit error rate (BER) used as a basis for determining when to reallocate a channel has been chosen as 10^{-3}. Therefore, the acceptable number of parity errors in a one second interval that do not exceed 10^{-3} must be determined. To establish the acceptable parity errors, the probable number of frame errors represented by each observed (monitored) parity error must be predicted. Given the number of monitored parity errors, the probable number of frame errors per monitored parity error, and the number of bit errors represented by a frame (parity) error, a probable bit error rate can be derived.

A statistical technique is used and the following assumptions are made:

1. Errors have a Poisson distribution, and
2. If the number of monitored parity errors is small (< 10) with respect to the total number of "samples" (100), the monitored parity error rate (MPER) reflects the mean frame error rate (FER).

Since a monitored parity error (MPE) represents 80 frames, assumption 2 implies that the number of frame errors (FEs) "behind" each parity error is equal to 80 MPER. That is, for 100 parity samples at 10 ms per sample, the mean number of frame errors per parity error is equal to 0.8 times the count of MPEs in one second. For example, if 3 MPEs are observed in a one second period, the mean number of FEs for each MPE is 2.4. Multiplying the desired bit error rate times the sample size and dividing by the bit errors per frame error yields the equivalent number of frame errors in the sample. The number of FEs is also equal to the product of the number of MPEs and the number of FEs per MPE. Given the desired BER, a solution set for the following equation can be determined.

$$(MPE\frac{FE}{MPE})=0.8MPE$$

The Poisson distribution, as follows, is used to compute the probability of a given number of FEs represented by a MPE (χ), and assumption 2, above, is used to arrive at the mean number of FEs per MPE (μ).

$$P(x)=\frac{e^{-\mu}\mu^{x}}{x!}$$

Since the desired bit error rate is a maximum, the Poisson equation is applied successively with values for χ of 0 up to the maximum number. The sum of these probabilities is the probability that no more than χ frame errors occurred for each monitored parity error.

The results for a bit error rate of 10^{-3} and bit errors per frame error of 1 and 8 are shown in Table 2.

Table 2: Bit Error Rate Probability

Bit Errors per Frame Error	Monitored Parity Errors	Maximum Frame Errors/ Monitored Parity Error (x)	Average Frame Errors/ Monitored Parity Error (μ)	Probability of BER $\leq 10^{-3}$
	2	4	1.6	98%
8	3	3	2.4	78%
	4	2	3.2	38%
	8	8	6.4	80%
1	9	7	7.2	56%
	10	7	8.0	45%

Using this technique, a value of 4 monitored parity errors detected during a one second integration was determined as the threshold to reallocate service of an ISU to a new channel. This result is arrived at by assuming a worst case of 8 bit errors per frame error, but a probability of only 38% that the bit error rate is better than 10^{-3}. The product of the bit errors per frame, monitored parity errors and maximum frame errors per monitored parity error must be 64, for a bit error rate of 10^{-3} (64 errors in 64k bits). Therefore, when the sampling of the parity errors in the error timer event is four or greater, the channel allocator is notified of a corrupted channel. If the sampled monitored parity errors is less than 4, the interrupt priority is lowered below parity and the error timer event is disabled, ending the timer error event

and the channels are then monitored as shown in the flow diagram of 27A.

The following is a description of the long integration operation performed by the background monitor routine (Figure 30) of the channel monitor 296. The background monitor routine is used to ensure quality integrity for channels requiring greater quality than the short integration 10^{-3} bit error rate. As the flow diagram shows in Figure 30, the background monitor routine operates over a specified time for each service type, updates the channel quality database table 300, clears the background count, determines if the integrated errors exceed the allowable limits determined for each service type, and notifies the channel allocator 304 of bad channels as needed.

In operation, on one second intervals, the background monitor updates the channel quality database table. Updating the channel quality data table has two purposes. The first purpose is to adjust the bit error rate and number of errored seconds data of error free channels to reflect their increasing quality. The second purpose is to integrate intermittent errors on monitored channels which are experiencing error levels too low to result in short integration time reallocation (less than 4 parity errors/second). Channels in this category have their BER and numbers of errored seconds data adjusted, and based on the data, may be re-allocated. This is known as long integration time re-allocation, and the default criteria for long integration time re-allocation for each service type are shown as follows:

Table 3

Service type:	Maximum BER:	Integration Time:	Errored seconds	Monitoring Period:
POTS	10^{-3}	1 second		
ISDN	10^{-6}	157 seconds	8 %	1 hour
DDS	10^{-7}	157 seconds	0.5 %	1 hour
DS1	10^{-9}	15.625 seconds	0.04 %	7 hours

Because POTS service does not require higher quality than 10^{-3}, corrupted channels are sufficiently eliminated using the short integration technique and

long integration is not required.

As one example of long integration for a service type, the background monitor shall be described with reference to a channel used for ISDN transport. Maximum bit error rate for the channel may be 10^{-6}, the number of seconds utilized for integration time is 157, the maximum number of errored seconds allowable is 8% of the 157 seconds, and the monitoring period is one hour. Therefore, if the summation of errored seconds is greater than 8% over the 157 seconds in any one hour monitoring period, the channel allocator 304 is notified of a bad channel for ISDN transport.

Unallocated or unused channels, but initialized and activated, whether used for reallocation for non-concentration services such as TR-8 or used for allocation or reallocation for concentration services such as TR-303, must also be monitored to insure that they are not bad, thereby reducing the chance that a bad channel will be allocated or reallocated to an ISU 100. To monitor unallocated channels, channel monitor 304 uses a backup manager routine (Figure 31) to set up unallocated channels in a loop in order to accumulate error data used to make allocation or re-allocation decisions. When an unallocated channel experiences errors, it will not be allocated to an ISU 100 for one hour. After the channel has remained idle (unallocated) for one hour, the channel monitor places the channel in a loop back mode to see if the channel has improved. In loop back mode, the CXMU 56 commands an initialized and activated ISU 100 to transmit a message on the channel long enough to perform short or long integration on the parity errors as appropriate. In the loop back mode, it can be determined whether the previously corrupted channel has improved over time and the channel quality database is updated accordingly. When not in the loop back mode, such channels can be powered down.

As described above, the channel quality database includes information to allow a reallocation or allocation to be made in such a manner that the channel used for allocation or reallocation is not corrupted. In addition, the information of the channel quality database can be utilized to rank the unallocated channels as for quality such that they can be allocated effectively.

For example, a channel may be good enough for POTS and not good enough for ISDN. Another additional channel may be good enough for both. The additional channel may be held for ISDN transmission and not used for POTS. In addition, a particular standby channel of very good quality may be set aside such that when ingress is considerably high, one channel is always available to be switched to.

In addition, an estimate of signal to noise ratio can also be determined for both unallocated and allocated channels utilizing the equalizer 214 of the MCC modem 82 upstream receiver architecture as shown in Figure 15. As described earlier, the equalizer was previously utilized to determine whether a channel was idle such that it could be allocated. During operation of the equalizer, as previously described, an error is generated to update the equalizer coefficients. The magnitude of the error can be mapped into an estimate of signal to noise ratio (SNR) by signal to noise monitor 305 (Figure 15). Likewise, an unused channel should have no signal in the band. Therefore, by looking at the variance of the detected signal within the unused FFT bin, an estimate of signal to noise ratio can be determined. As the signal to noise ratio estimate is directly related to a probable bit error rate, such probable bit error rate can be utilized for channel monitoring in order to determine whether a bad or good channel exists.

Therefore, for reallocation for nonconcentration services such as TR-8 services, reallocation can be performed to unallocated channels with such unallocated channels monitored through the loopback mode or by SNR estimation by utilization of the equalizer. Likewise, allocation or reallocation for concentration services such as TR-303 services can be made to unallocated channels based upon the quality of such unallocated channels as determined by the SNR estimation by use of the equalizer.

With respect to channel allocation, a channel allocator routine for channel allocator 304 examines the channel quality database table to determine which DS0+ channels to allocate to an ISU 100 for a requested service. The channel allocator also checks the status of the ISU and channel units to verify in-service status and proper type for the requested service. The channel

allocator attempts to maintain an optimal distribution of the bandwidth at the ISUs to permit flexibility for channel reallocation. Since it is preferred that ISUs 100, at least HISUs, be able to access only a portion of the RF band at any given time, the channel allocator must distribute channel usage over the ISUs so as to not overload any one section of bandwidth and avoid reallocating in-service channels to make room for additional channels.

The process used by the channel allocator 304 is to allocate equal numbers of each ISU type to each band of channels of the 6 MHz spectrum. If necessary, in use channels on an ISU can be moved to a new band, if the current ISU band is full and a new service is assigned to the ISU. Likewise, if a channel used by an ISU in one band gets corrupted, the ISU can be reallocated to a channel in another subband or band of channels. Remember that the distributed IOC channels continue to allow communication between the HDT 12 and the HISUs as an HISU always sees one of the IOC channels distributed throughout the spectrum. Generally, channels with the longest low-error rate history will be used first. In this way, channels which have been marked bad and subsequently reallocated for monitoring purposes will be used last, since their histories will be shorter than channels which have been operating in a low error condition for a longer period.

Second Embodiment of Telephony Transport System

A second embodiment of an OFDM telephony transport system, referring to Figures 24-27 shall be described. The 6 MHz spectrum allocation is shown in Figure 24. The 6 MHz bandwidth is divided into nine channel bands corresponding to the nine individual modems 226 (Figure 25). It will be recognized by one skilled in the art that less modems could be used by combining identical operations. Each of the channel bands includes 32 channels modulated with a quadrature 32-ary format (32-QAM) having five bits per symbol. A single channel is allocated to support transfer of operations and control data (IOC control data) for communication between an HDT 12 and ISUs 100. This channel uses BPSK modulation.

The transport architecture shall first be described with respect to

downstream transmission and then with respect to upstream transmission. Referring to Figure 25, the MCC modem 82 architecture of the HDT 12 will be described. In the downstream direction, serial telephony information and control data is applied from the CXMC 80 through the serial interface 236. The serial data is demultiplexed by demultiplexer 238 into parallel data streams. These data streams are submitted to a bank of 32 channel modems 226 which perform symbol mapping and fast fourier transform (FFT) functions. The 32 channel modems output time domain samples which pass through a set of mixers 240 that are driven by the synthesizer 230. The mixers 240 create a set of frequency bands that are orthogonal, and each band is then filtered through the filter/combiner 228. The aggregate output of the filter/combiner 228 is then upconverted by synthesizer 242 and mixer 241 to the final transmitter frequency. The signal is then filtered by filter 232, amplified by amplifier 234, and filtered again by filter 232 to take off any noise content. The signal is then coupled onto the HFC distribution network via telephony transmitter 14.

On the downstream end of the HFC distribution network 11, an ISU 100 includes a subscriber modem 258 as shown in Figure 26. The downstream signals are received from the ODN 18 through the coax leg 30, and are filtered by filter 260 which provides selectivity for the entire 6 MHz band. Then the signal is split into two parts. The first part provides control data and timing information to synchronize clocks for the system. The second part provides the telephony data. With the control data received separately from the telephony data, this is referred to as previously described above as an out of band ISU. The out of band control channel which is BPSK modulated is split off and mixed to baseband by mixer 262. The signal is then filtered by filter 263 and passed through an automatic gain control stage 264 and a Costas loop 266 where carrier phase is recovered. The signal that results is passed into a timing loop 268 so timing can be recovered for the entire modem. The IOC control data, which is a byproduct of the Costas loop, is passed into the 32 channel OFDM modem 224 of the ISU 100. The second part of the downstream OFDM waveform is mixed to base band by mixer 270

and associated synthesizer 272. The output of the mixer 270 is filtered by filter 273 and goes through a gain control stage 274 to prepare it for reception. It then passes into the 32 channel OFDM modem 224.

Referring to Figure 27, the IOC control data is hard limited through function block 276 and provided to microprocessor 226. The OFDM telephony data is passed through an analog to digital converter 278 and input to a first-in first-out buffer 280 where it is stored. When a sufficient amount of information is stored, it is accepted by the microprocessor 226 where the remainder of the demodulation process, including application of an FFT, takes place. The microprocessor 226 provides the received data to the rest of the system through the receive data and receive data clock interface. The fast fourier transform (FFT) engine 282 is implemented off the microprocessor. However, one skilled in the art will recognize that the FFT 282 could be done by the microprocessor 226.

In the upstream direction, data enters the 32 channel OFDM modem 224 through the transmit data ports and is converted to symbols by the microprocessor 226. These symbols pass through the FFT engine 282, and the resulting time domain waveform, including guard samples, goes through a complex mixer 284. The complex mixer 284 mixes the waveform up in frequency and the signal is then passed through a random access memory digital to analog converter 286 (RAM DAC). The RAM DAC contains some RAM to store up samples before being applied to the analog portion of the ISU upstream transmitter (Figure 26). Referring to Figure 26, the OFDM output for upstream transport is filtered by filter 288. The waveform then passes through mixer 290 where it is mixed under control of synthesizer 291 up to the transmit frequency. The signal is then passed through a processor gain control 292 so that amplitude leveling can take place in the upstream path. The upstream signal is finally passed through a 6 MHz filter 294 as a final selectivity before upstream transmission on the coaxial leg 30 to the ODN 18.

In the upstream direction at the HDT 12 side, a signal received on the coax from the telephony receiver 16 is filtered by filter 244 and amplified by

amplifier 246. The received signal, which is orthogonally frequency division multiplexed, is mixed to baseband by bank of mixers 248 and associated synthesizer 250. Each output of the mixers 248 is then filtered by baseband filter bank 252 and each output time domain waveform is sent then to a demodulator of the 32 channel OFDM modems 226. The signals pass through a FFT and the symbols are mapped back into bits. The bits are then multiplexed together by multiplexer 254 and applied to CXMC 56 through the other serial interface 256.

As shown in this embodiment, the ISU is an out of band ISU as utilization of separate receivers for control data and telephony data is indicative thereof as previously discussed. In addition, the separation of the spectrum into channel bands is further shown. Various other embodiments as contemplated by the accompanying claims of the transport system are possible by building on the embodiments described herein. In one embodiment, an IOC control channel for at least synchronization information transport, and the telephony service channels or paths are provided into a single format. The IOC link between the HDT 12 and the ISUs 100 may be implemented as four BPSK modulated carriers operating at 16 kbps, yielding a data rate of 64 kbps in total. Each subscriber would implement a simple separate transceiver, like in the second embodiment, which continuously monitors the service channel assigned to it on the downstream link separately from the telephony channels. This transceiver would require a tuned oscillator to tune to the service IOC channel. Likewise, an IOC channel could be provided for channel bands of the 6 MHz bandwidth and the channel bands may include orthogonal carriers for telephony data and an IOC channel that is received separately from the reception of the orthogonal carriers.

In another embodiment, instead of 4 BPSK channels, a single 64 kbps IOC channel is provided. This single channel lies on the OFDM frequency structure, although the symbol rate is not compatible with the telephony symbol rate of OFDM framework. This single wide band signal requires a wider band receiver at the ISU 100 such that the IOC link between the HDT 12 and ISUs is always possible. With single channel support it is possible to

use a fixed reference oscillator that does not have to tune across any part of the band in the subscriber units. However, unlike in the first embodiment where the IOC channels are distributed across the spectrum allowing for narrow band receivers, the power requirements for this embodiment would increase because of the use of the wide band receiver at the ISU 100.

In yet another embodiment, the IOC link may include two IOC channels in each of 32 OFDM channel groups. This increases the number of OFDM carriers to 34 from 32 in each group. Each channel group would consist of 34 OFDM channels and a channel band may contain 8 to 10 channels groups. This approach allows a narrow band receiver to be used to lock to the reference parameters provided by the HDT 12 to utilize an OFDM waveform, but adds the complexity of also having to provide the control or service information in the OFDM data path format. Because the subscriber could tune to any one of the channel groups, the information that is embedded in the extra carriers must also be tracked by the central office. Since the system needs to support a timing acquisition requirement, this embodiment may also require that a synchronization signal be place off the end of the OFDM waveform.

It is to be understood, however, that even though numerous characteristics of the present invention have been set forth in the foregoing description, together with details of the structure and function of the invention, the disclosure is illustrative and changes in matters of order, shape, size, and arrangement of the parts, and various properties of the operation may be made within the principles of the invention and to the full extent indicated by the broad general meaning of the terms in which the appended claims are expressed.

WHAT IS CLAIMED IS:

1. A method for monitoring at least one telephony communication n-bit channel wherein one of the bits being a parity bit, the method comprising the steps of:

sampling the parity bit of the n-bit channel; and

deriving a probable bit error rate from the sampling of the parity bit.

2. The method of claim 1, further comprising the step of periodically monitoring and accumulating error data for at least one unallocated telephony communication channel.

3. The method of claim 1, further comprising the steps of:

comparing the probable bit error rate to a pre-determined bit error rate value to determine if the at least one telephony communication n-bit channel is corrupted; and

re-allocating the at least one telephony communication n-bit channel to an uncorrupted and unallocated telephony communication n-bit channel, if the at least one telephony communication n-bit channel is corrupted.

4. The method of claim 1 further comprising the steps of:

comparing the probable bit error rate to a pre-determined bit error rate value to determine if the at least one telephony communication n-bit channel is corrupted; and

increasing transmission power of the n-bit channel if the n-bit channel is corrupted, while maintaining total system power.

5. A method for monitoring at least one telephony communication n-bit channel wherein one of the bits being a parity bit, the method comprising the steps of:

sampling the parity bit of the n-bit channel;

deriving a probable bit error rate from the sampling of the parity bit over a time period; and

comparing the probable bit error rate over the time period to a pre-determined bit error rate value to determine if the n-bit channel is corrupted.

6. The method of claim 5 further comprising the step of re-allocating the communication from the n-bit channel to a different n-bit channel based on the comparison.

7. The method of claim 6 wherein the at least one telephony communication n-bit channel is contained within a band of a plurality of telephony communication n-bit channels, the band being associated with at least one control channel, and further wherein the different n-bit channel is located within the band.

8. The method of claim 6 wherein the at least one telephony communication n-bit channel is contained within a band of a plurality of telephony communication n-bit channels, the band being associated with at least one control channel, and further wherein the different n-bit channel is located in a second band of a plurality of telephony communication n-bit channels having another at least one control channel associated therewith.

9. The method of claim 5 further comprising the step of increasing transmission power of the n-bit channel if the n-bit channel is corrupted, while maintaining total system power.

10. The method of claim 5 further comprising the step of storing the probable bit error rate in a table, wherein the table can be used for allocating future communications on an n-bit channel.

11. The method of claim 5 further comprising the steps of:

deriving at least one additional probable bit error rate from the sampling of the parity bit over at least one longer time period if the channel is not corrupted: and

comparing the at least one additional probable bit error rate to an additional pre-determined bit error rate value to determine if the n-bit channel is corrupted.

12. The method of claim 11 wherein the predetermined bit error rate value is for a telephony communication service and the additional predetermined bit error rate value is for an additional telephony communication service.

13. The method of claim 12 wherein one of the telephony communication services is ISDN.

14. The method of claim 11 further comprising the step of increasing transmission power of the n-bit channel if the n-bit channel is corrupted, while maintaining total system power.

15. The method of claim 11 further comprising the step of re-allocating the communication from the n-bit channel to a different n-bit channel based on the comparison of the at least one additional probable bit error rate to an additional pre-determined bit error rate value.

16. A method for monitoring at least one telephony communication n-bit channel wherein one of the bits being a parity bit, the method comprising the steps of:

sampling the parity bit of the n-bit channel over a first time period;

deriving a probable bit error rate from the sampling of the parity bit over the first time period;

comparing the probable bit error rate over the first time period to a pre-determined bit error rate value to determine if the n-bit channel is corrupted; and

accumulating a probable bit error rate over a plurality of successive time periods if the n-bit channel is not corrupted.

17. The method of claim 16 further comprising the step of comparing the accumulated probable bit error rate over the successive time periods to at least one additional pre-determined bit error rate value to determine if the n-bit channel is corrupted.

18. The method of claim 17 further comprising the step of re-allocating communication from the n-bit channel to a second n-bit channel if the n-bit channel is corrupted.

19. The method of claim 17 further comprising the step of increasing transmission power of the n-bit channel if the n-bit channel is corrupted. while maintaining total system power.

20. The method of claim 19 wherein the predetermined bit error rate value is associated with a telephony communication service and the at least one additional predetermined bit error rate value is associated with at least one additional telephony communication service.

21. The method of claim 20 wherein one of the telephony communication services is ISDN.

22. The method of claim 16 further comprising the step of re-allocating communication from the n-bit channel to a second n-bit channel if the n-bit channel is corrupted.

23. The method of claim 16 further comprising the step of increasing transmission power of the n-bit channel if the n-bit channel is corrupted. while maintaining total system power.

24. A method for monitoring at least one telephony communication n-bit channel wherein one of the bits being a parity bit. the method comprising the steps of:

sampling the parity bit of the n-bit channel;

deriving a probable bit error rate from the sampling of the parity bit over a first time period;

comparing the probable bit error rate over the first time period to a first pre-determined bit error rate value to determine if the n-bit channel is corrupted;

deriving a probable bit error rate from the sampling of the parity bit over a second time period, the second time period being longer than the first time period and running concurrently with the first time period; and

comparing the probable bit error rate over the second time period to a second pre-determined bit error rate value to determine if the n-bit channel is corrupted.

25. The method of claim 24 further comprising the step of re-allocating the communication from the n-bit channel to a second n-bit channel if the n-bit channel is corrupted.

26. The method of claim 24 further comprising the step of increasing transmission power of the n-bit channel if the n-bit channel is corrupted, while maintaining total system power.

27. The method of claim 24 further comprising the step of storing the probable bit error rate in a table, wherein the table can be used for allocating future communications on an n-bit channel.

28. A method for monitoring at least one unallocated telephony communication channel, the method comprising the steps of:

periodically monitoring the at least one unallocated telephony communication channel;

accumulating error data for the at least one unallocated telephony communication channel; and

allowing the at least one unallocated telephony communication channel

to be allocated based on the error data.

29. The method of claim 28 further comprising the step of re-allocating a telephony communication from a corrupted telephony communication channel to the at least one unallocated telephony communication channel.

30. The method of claim 28, wherein the periodically monitoring the at least one unallocated telephony communication channel step includes:

transmitting an n-bit signal, wherein one of the bits being a parity bit, from the remote transmitter;

sampling the parity bit of the n-bit channel; and

deriving a probable bit error rate from the sampled parity bit.

31. The method of claim 28, wherein the unallocated channel is a powered-down allocated channel, the method further including the steps of:

powering up a remote transmitter at a remote location on the unallocated channel so that the channel can be monitored; and

powering down the remote transmitter after the channel is monitored..

32. The method of claim 28, further comprising the step of comparing the probable bit error rate to a pre-determined bit error rate value to determine if the channel is corrupt.

33. The method of claim 28 wherein the at least one unallocated telephony communication channel is one of a plurality of unallocated telephony communication channels, at least a certain number of the unallocated telephony communication channels being monitored; the method including the step of ranking a quality of at least a certain number of the unallocated channels based on such monitoring.

34. The method of claim 33 wherein the ranking step includes setting a high quality channel aside as a standby channel.

Figure 1

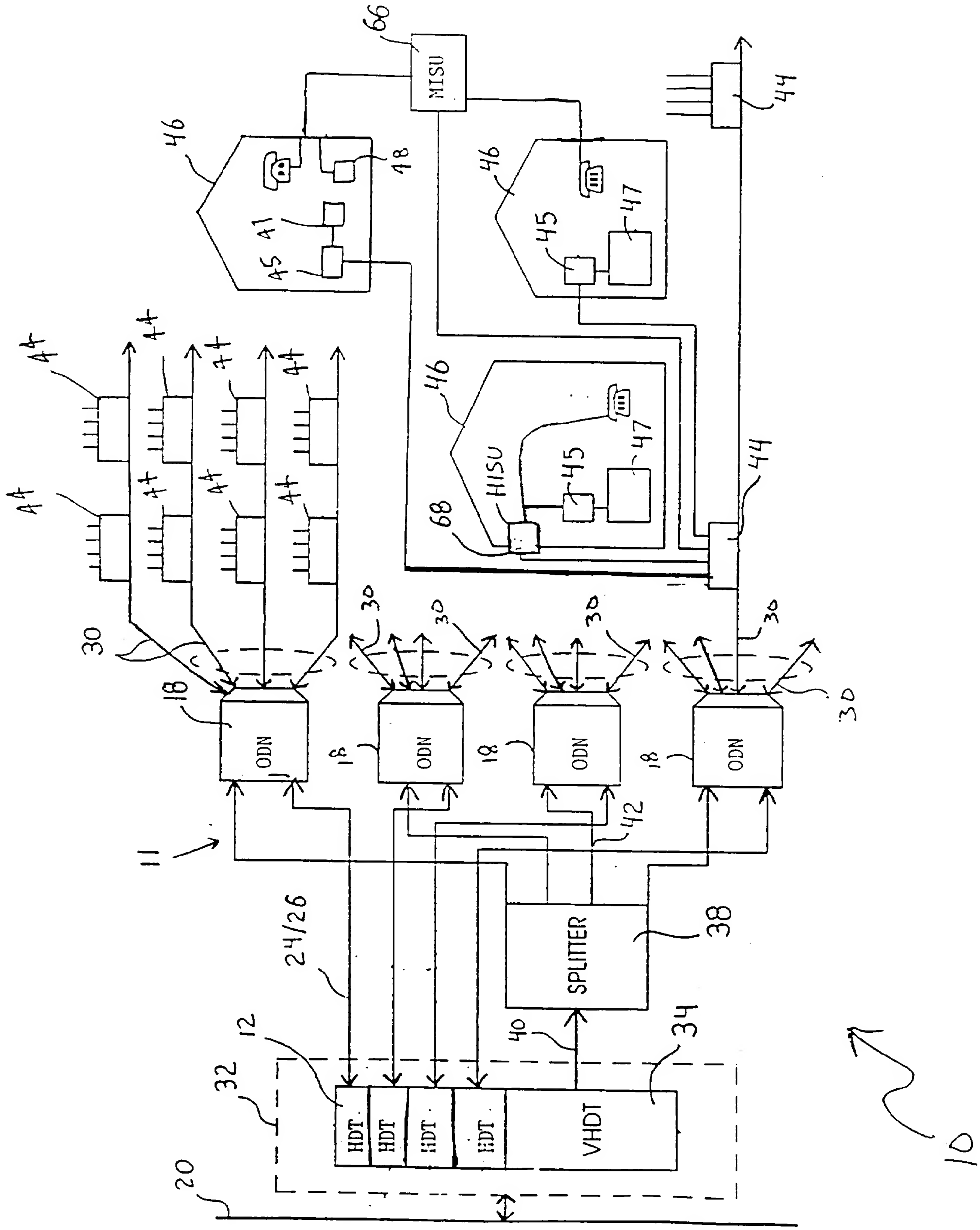
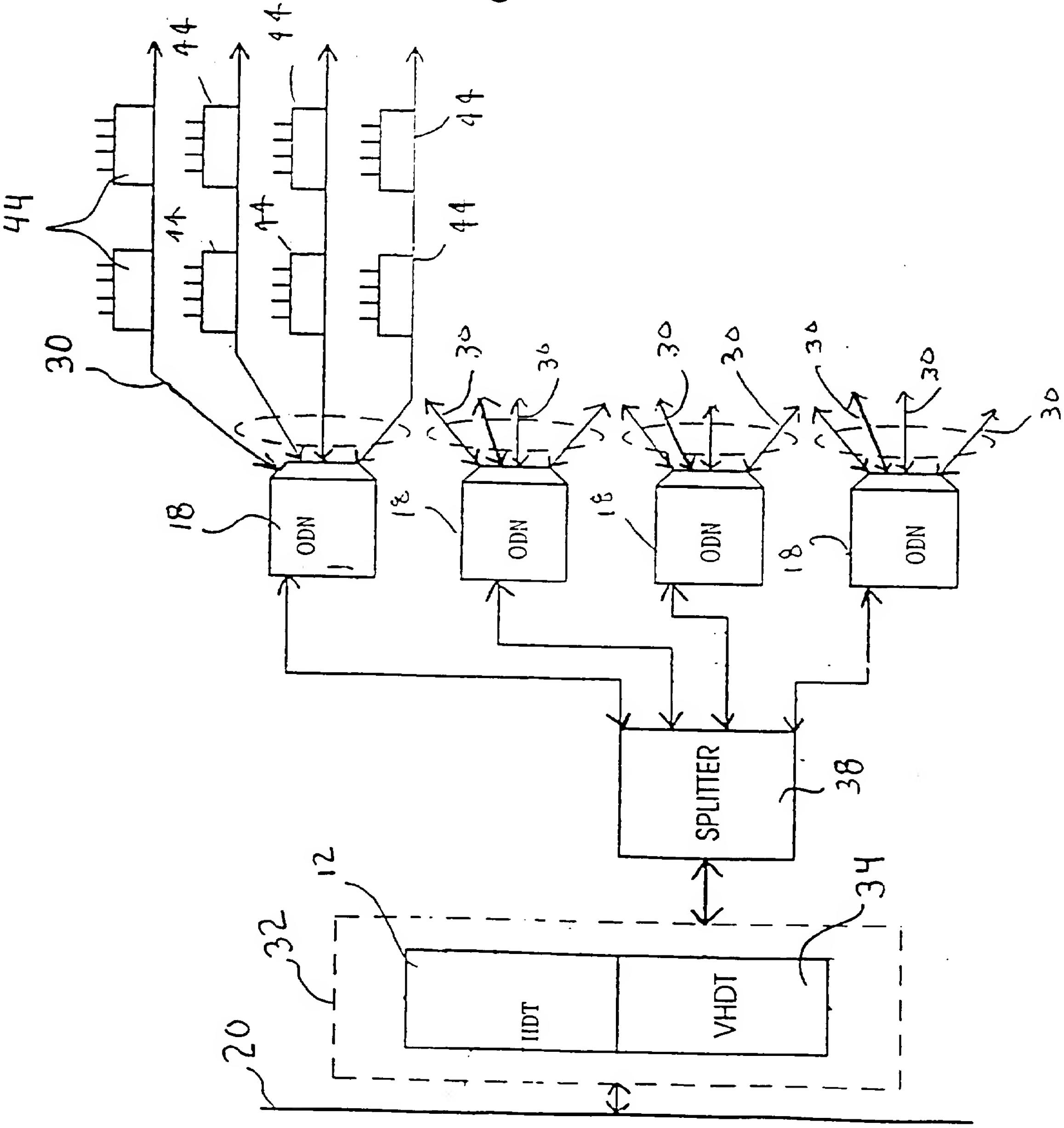


Figure 2



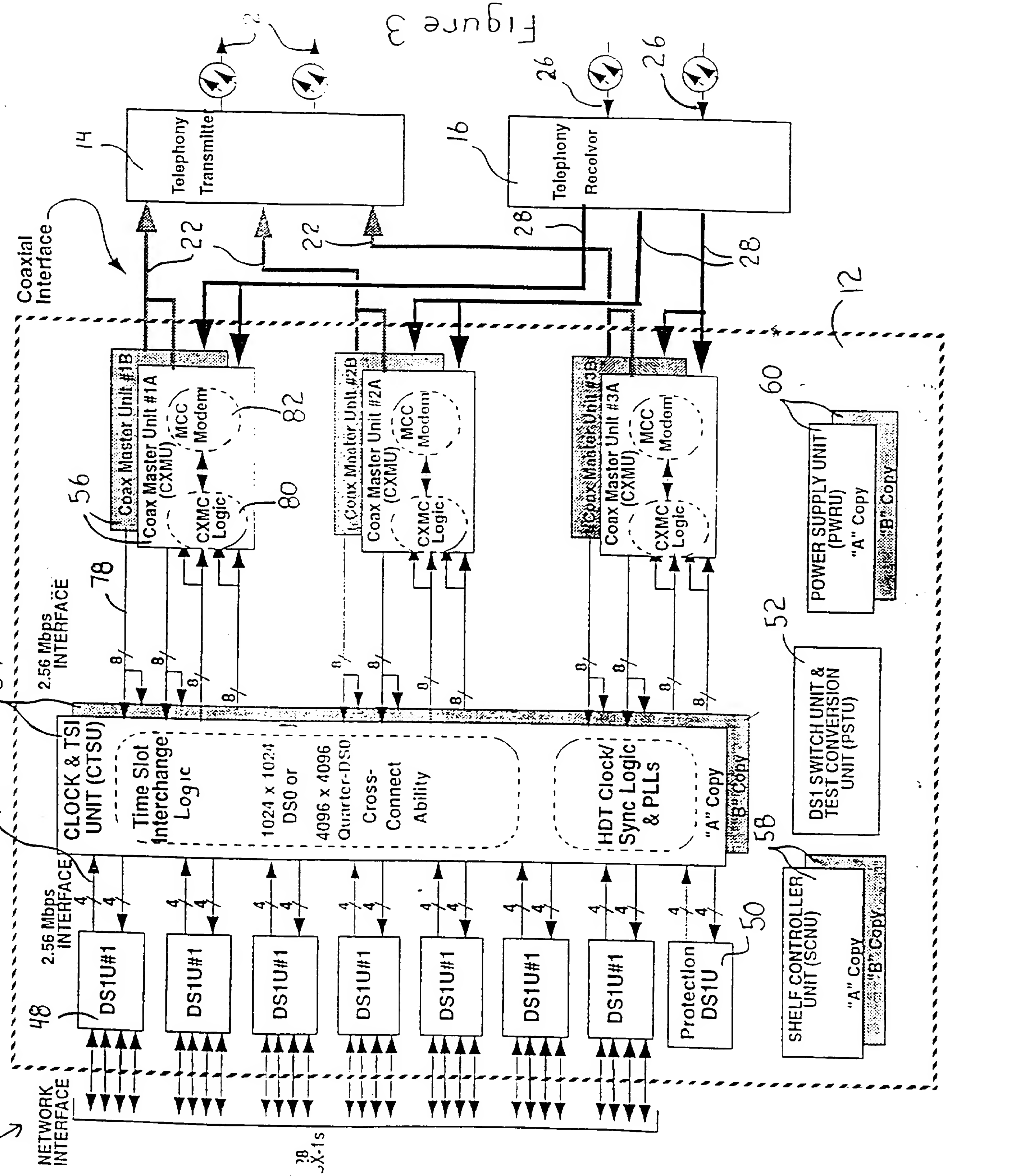


Figure 3
Telephony Transmitter
Telephony Receiver
Coaxial Interface
Coax Master Unit #1B
Coax Master Unit #1A (CXMU)
MCC Modem
CXMC Logic
Coax Master Unit #2B
Coax Master Unit #2A (CXMU)
Coax Master Unit #3B
Coax Master Unit #3A (CXMU)
POWER SUPPLY UNIT (PWRU)
"A" Copy
"B" Copy
2.56 Mbps INTERFACE
CLOCK & TSI UNIT (CTSU)
Time Slot Interchange Logic
1024 x 1024 DS0 or 4096 x 4096 Quarter-DS0 Cross-Connect Ability
HDT Clock/ Sync Logic & PLLs
DS1 SWITCH UNIT & TEST CONVERSION UNIT (PSTU)
SHELF CONTROLLER UNIT (SCNU)
DS1U#1
Protection DS1U
NETWORK INTERFACE
14
16
22
26
28
12
56
82
80
78
54
60
52
58
50
48
8
4

Figure 4

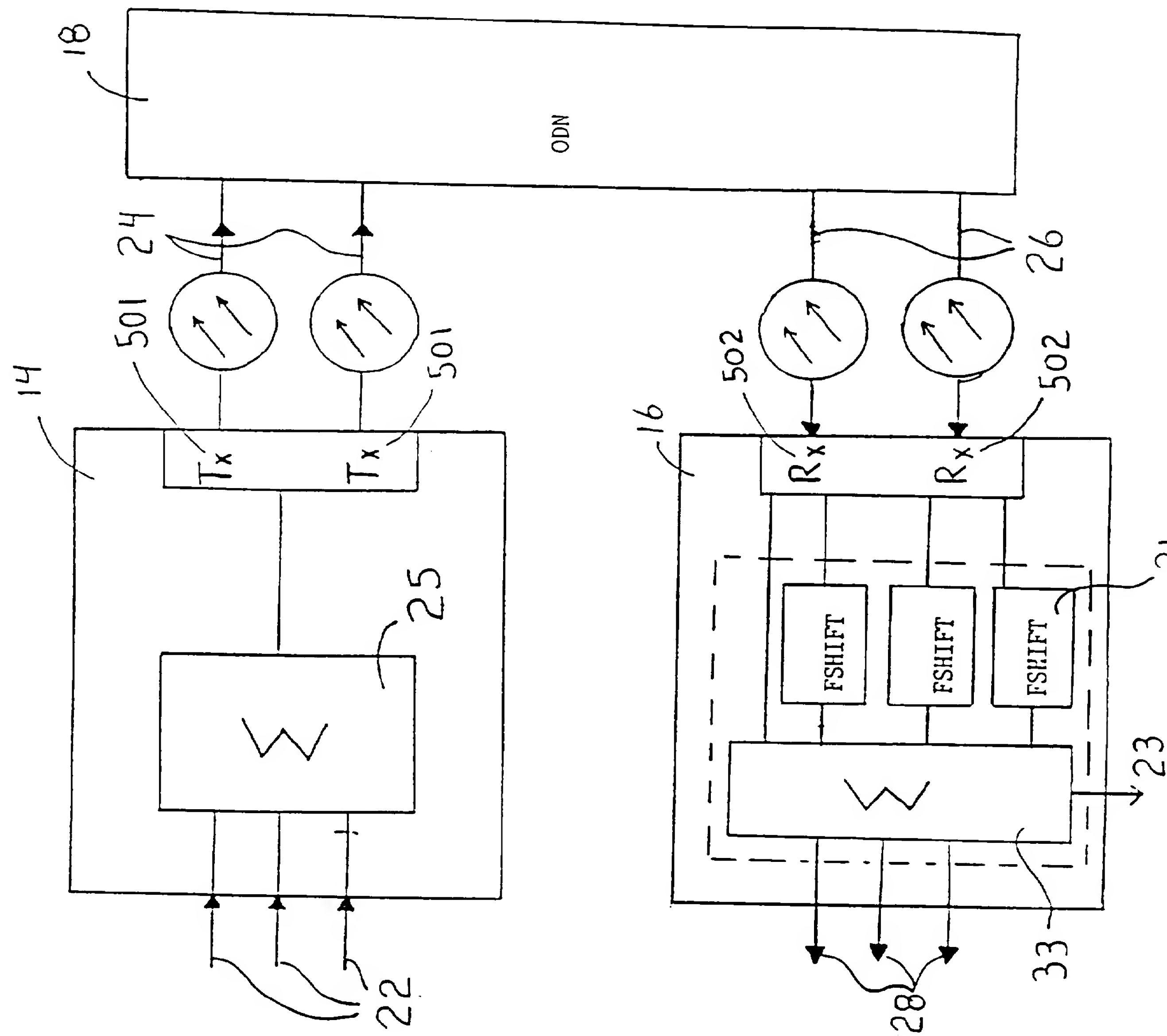

18
ODN
24
501
501
14
Tx
Tx
25
W
22
26
502
502
16
Rx
Rx
FSHIFT
FSHIFT
FSHIFT
31
W
23
33
28

Figure 5

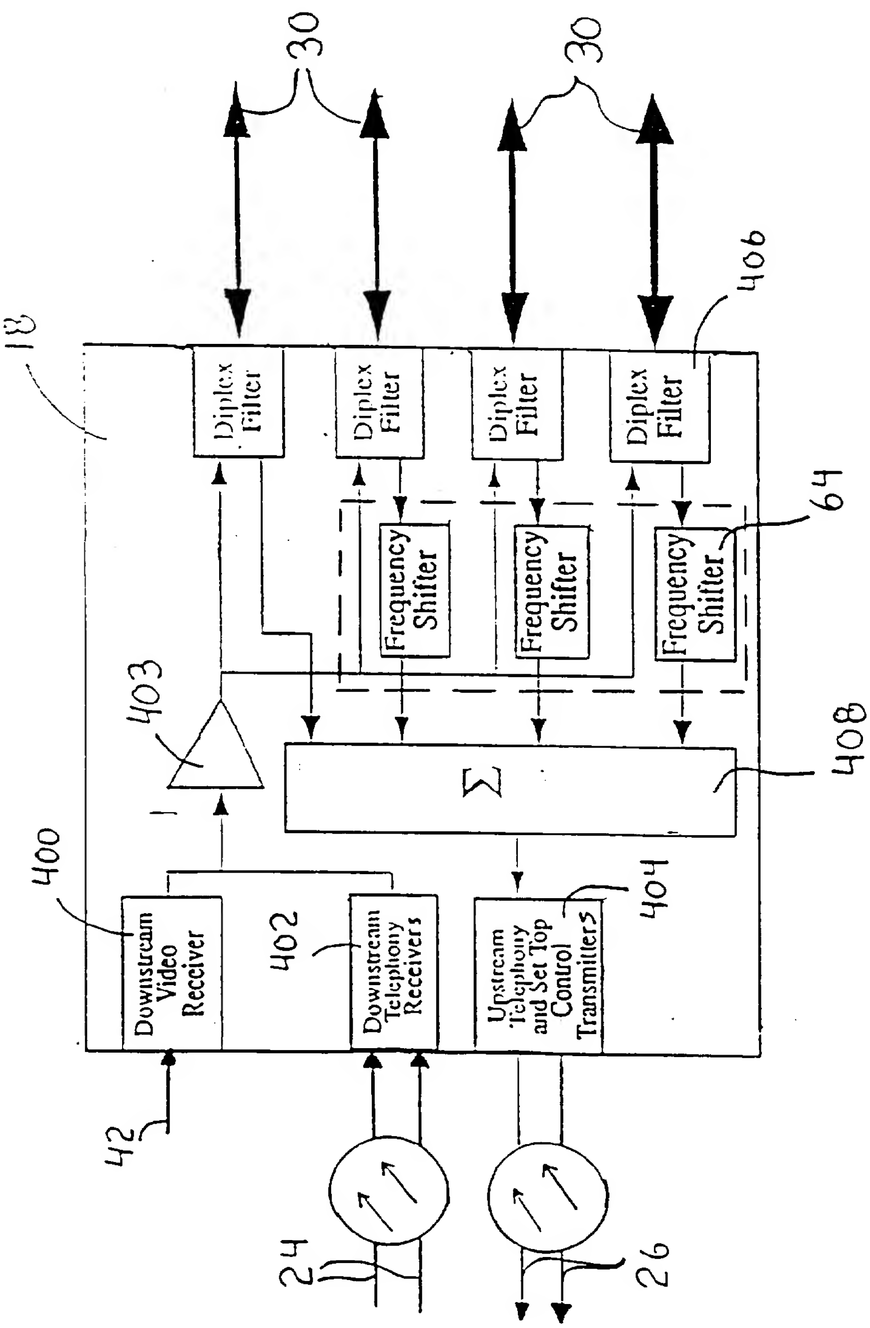

18
30
30
406
Diplex Filter
Diplex Filter
Diplex Filter
Diplex Filter
Frequency Shifter
Frequency Shifter
Frequency Shifter
64
403
Σ
408
400
Downstream Video Receiver
402
Downstream Telephony Receivers
Upstream Telephony and Set Top Control Transmitters
404
42
24
26

Figure 6

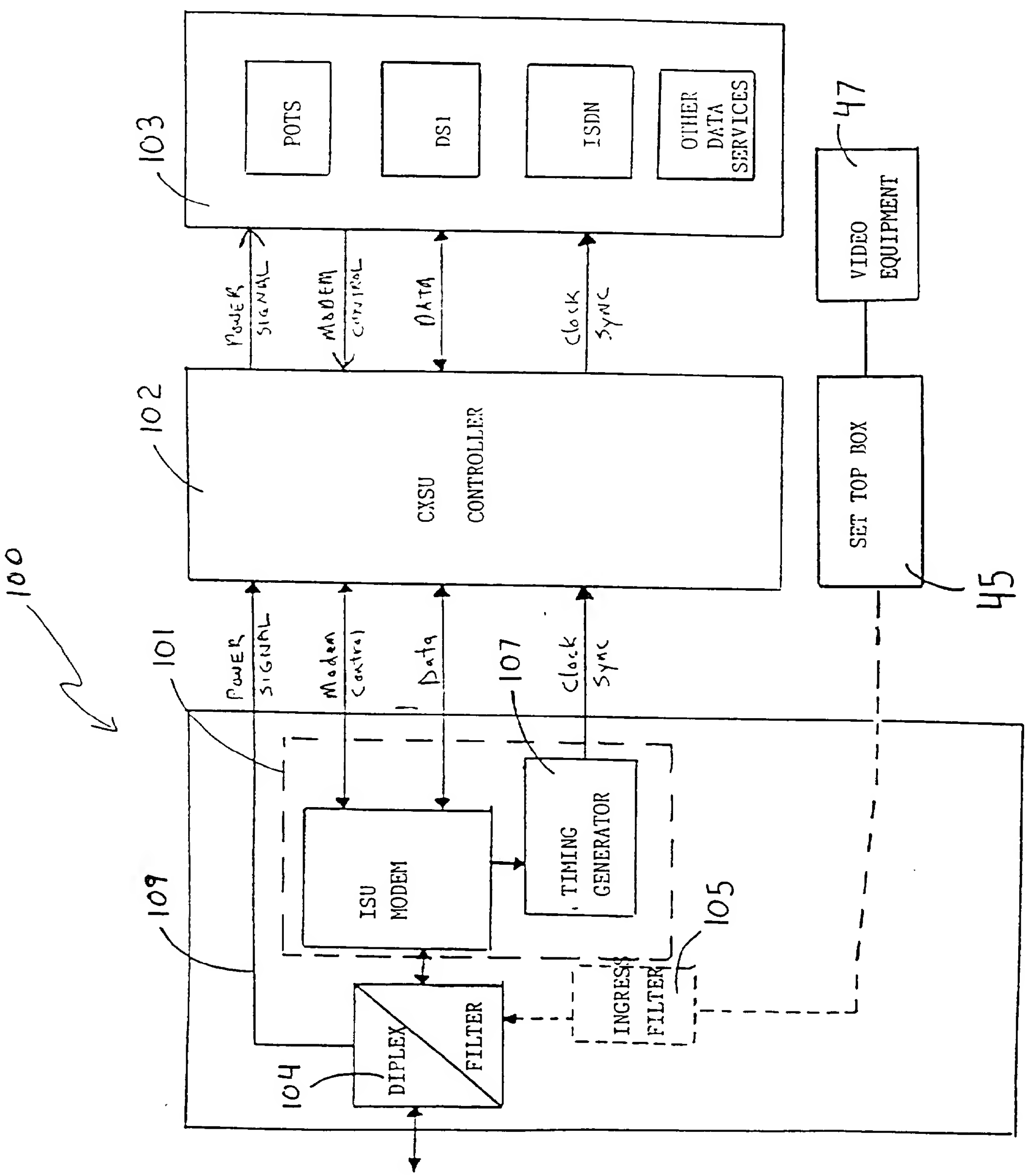
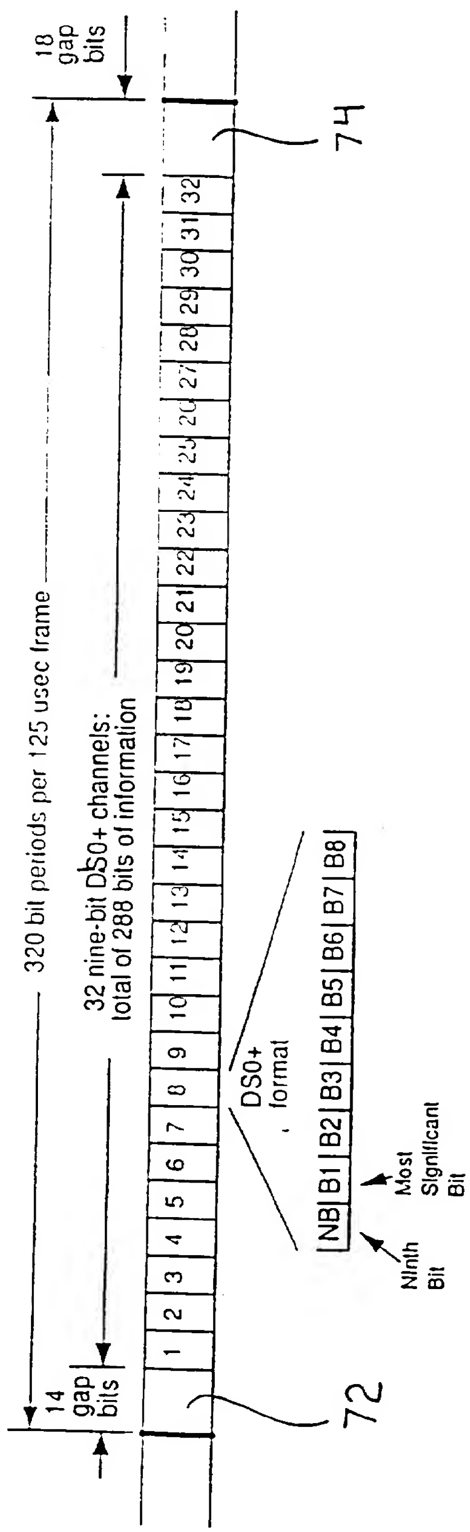

103
POTS
DS1
ISDN
OTHER DATA SERVICES
47
VIDEO EQUIPMENT
Power SIGNAL
Modem Control
Data
Clock Sync
102
CXSU CONTROLLER
SET TOP BOX
45
100
101
107
ISU MODEM
TIMING GENERATOR
109
105
INGRESS FILTER
DIPLEX FILTER
104

Figure 7A



320 bit periods per 125 usec frame
32 nine-bit DS0+ channels: total of 288 bits of information
18 gap bits
14 gap bits
1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31 32
74
72
DS0+ format
NB B1 B2 B3 B4 B5 B6 B7 B8
Ninth Bit
Most Significant Bit

Figure 7B

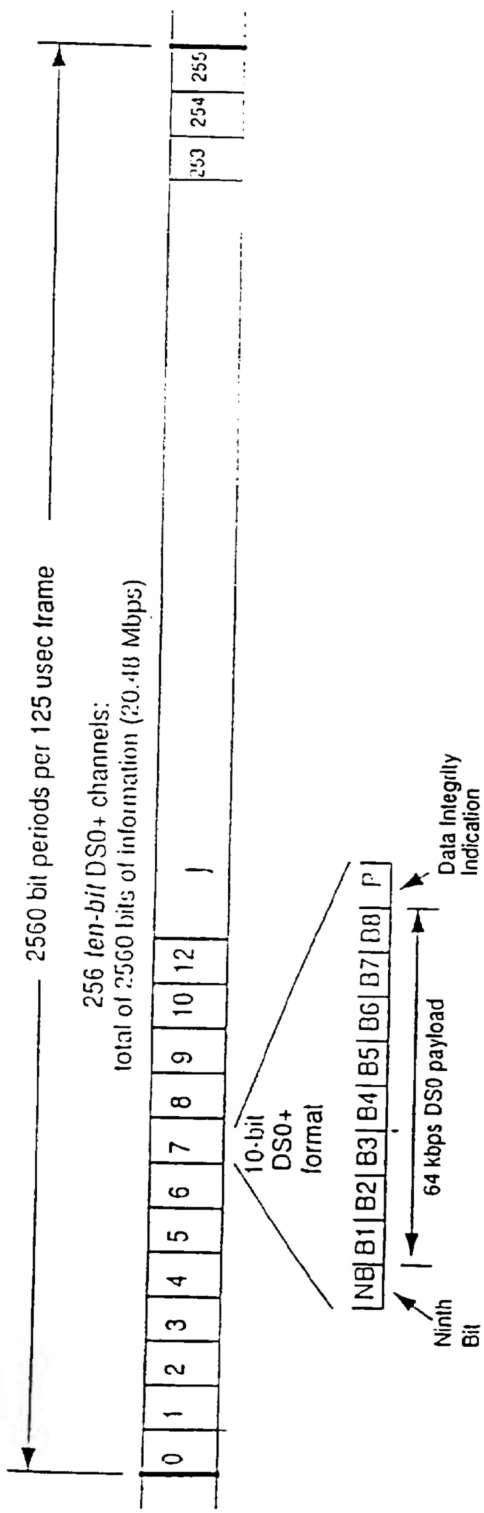

2560 bit periods per 125 usec frame
256 ten-bit DS0+ channels:
total of 2560 bits of information (20.48 Mbps)
0
1
2
3
4
5
6
7
8
9
10
12
253
254
255
10-bit DS0+ format
NB
B1
B2
B3
B4
B5
B6
B7
B8
P
Ninth Bit
64 kbps DS0 payload
Data Integrity Indication

Figure 7C

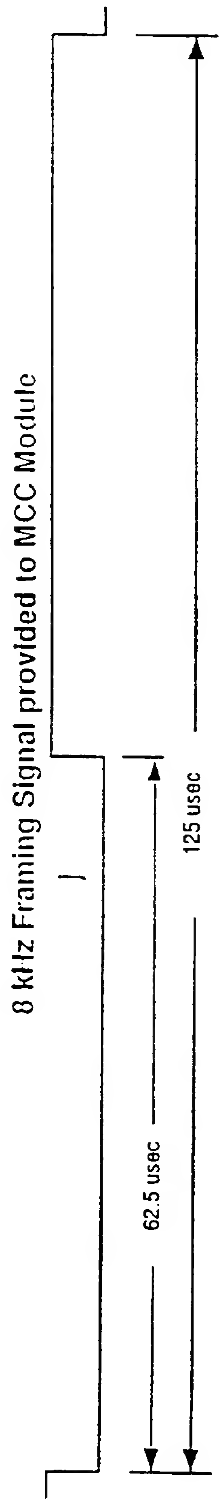
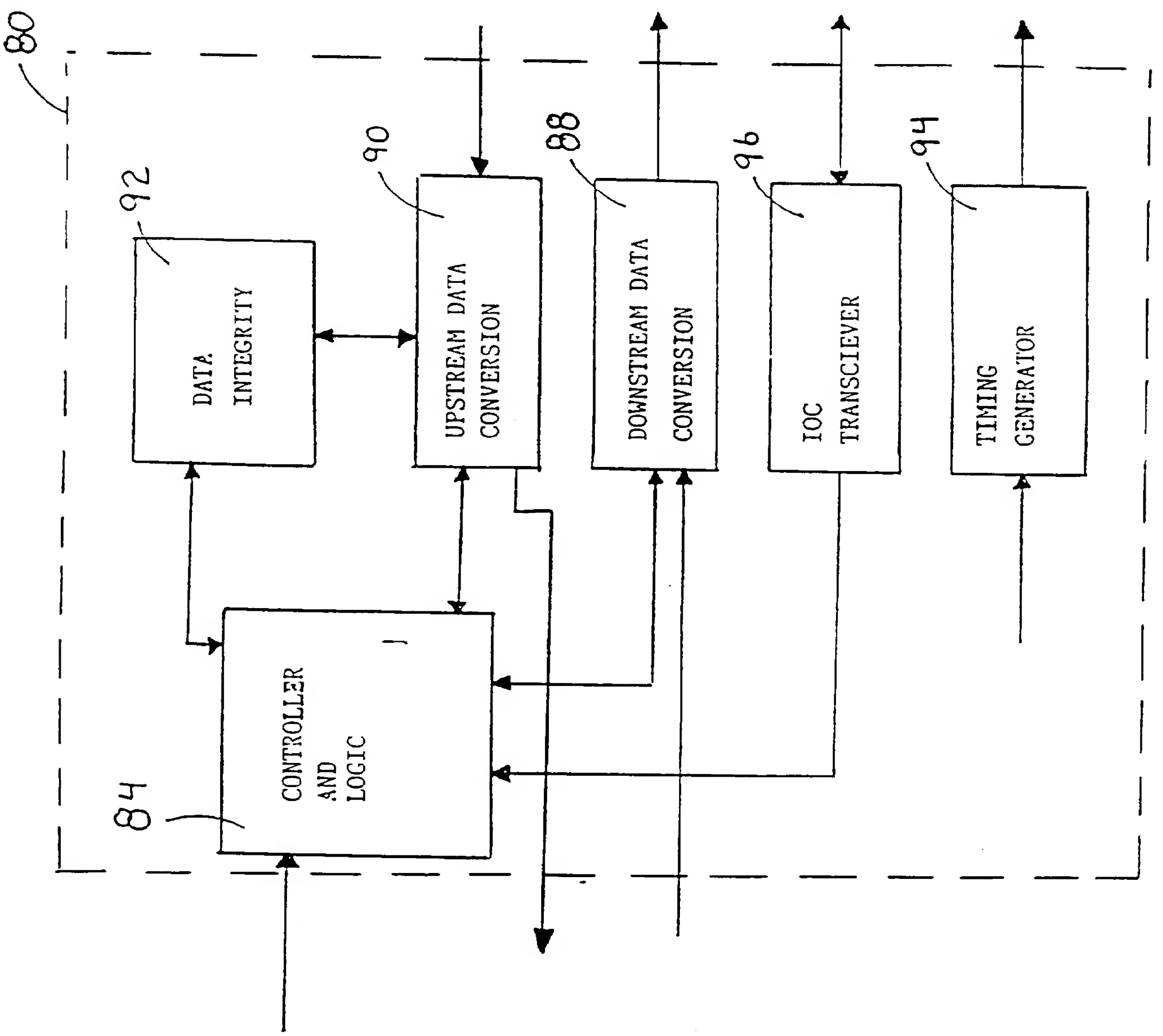
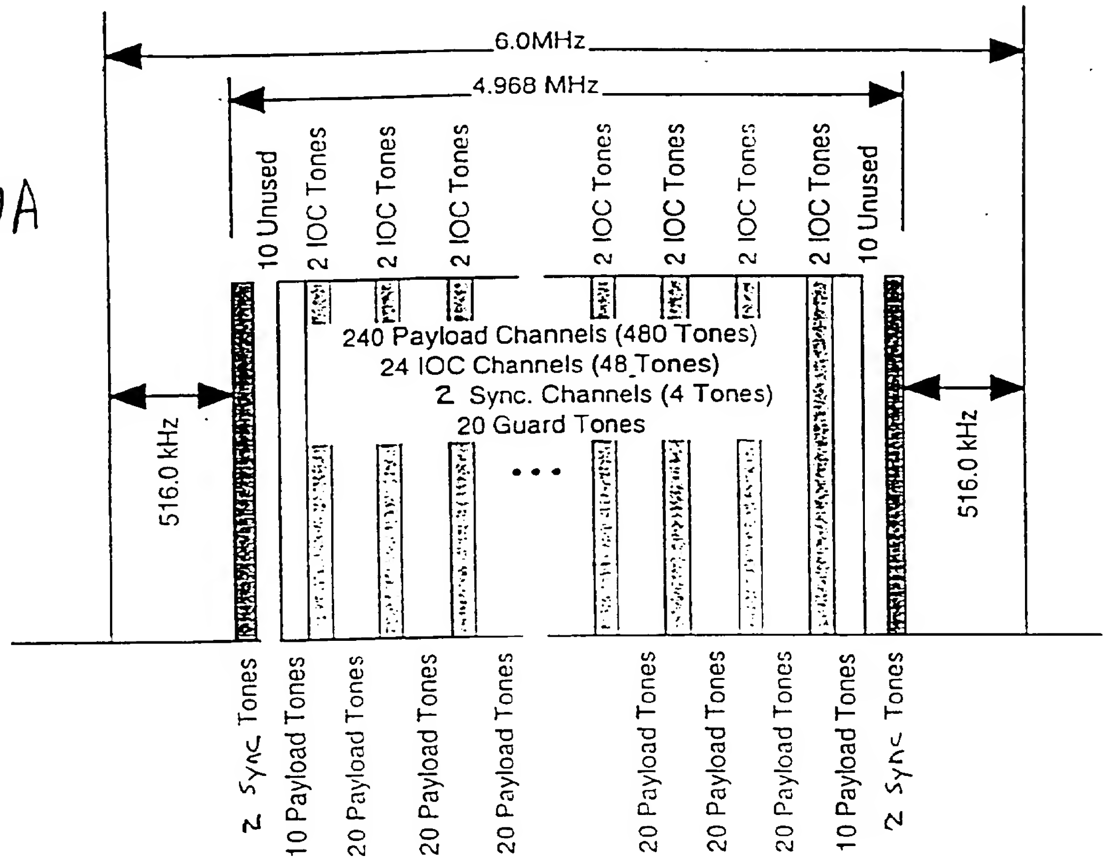

8 kHz Framing Signal provided to MCC Module
125 usec
62.5 usec

Figure 8

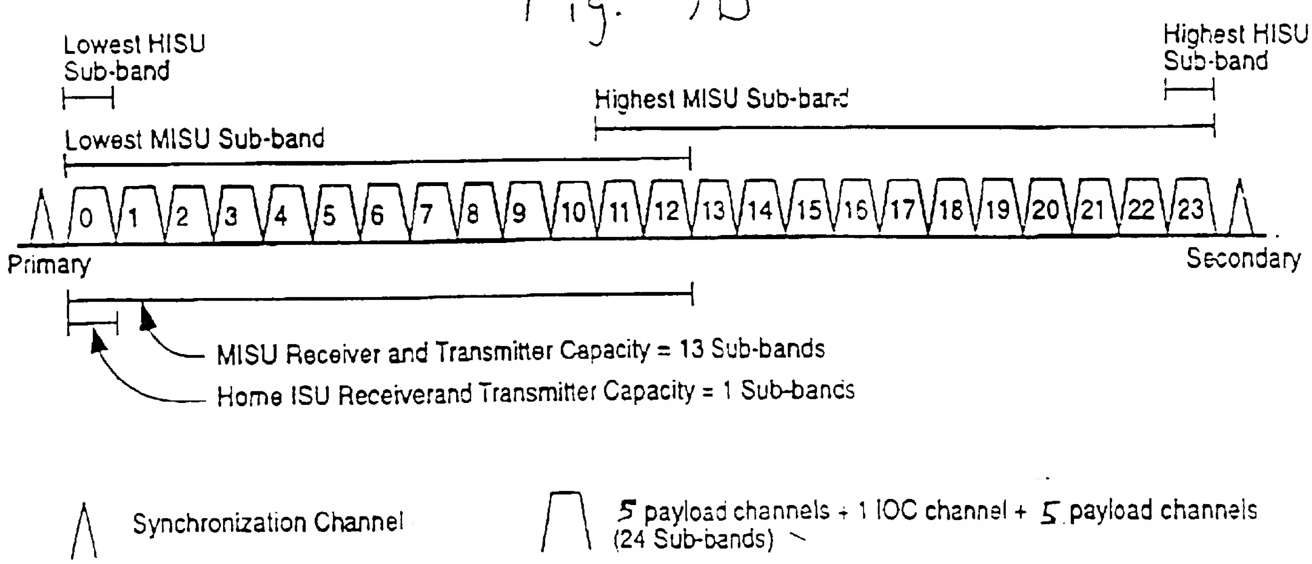

80
92
DATA INTEGRITY
90
UPSTREAM DATA CONVERSION
88
DOWNSTREAM DATA CONVERSION
96
IOC TRANSCIEVER
94
TIMING GENERATOR
84
CONTROLLER AND LOGIC



6.0MHz
4.968 MHz
10 Unused
2 IOC Tones
2 IOC Tones
2 IOC Tones
2 IOC Tones
2 IOC Tones
2 IOC Tones
2 IOC Tones
10 Unused
240 Payload Channels (480 Tones)
24 IOC Channels (48 Tones)
2 Sync. Channels (4 Tones)
20 Guard Tones
516.0 kHz
516.0 kHz
2 Sync Tones
10 Payload Tones
20 Payload Tones
20 Payload Tones
20 Payload Tones
20 Payload Tones
20 Payload Tones
20 Payload Tones
10 Payload Tones
2 Sync Tones

Fig. 9A

Fig. 9D



Lowest HISU Sub-band
Highest HISU Sub-band
Highest MISU Sub-band
Lowest MISU Sub-band
0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23
Primary
Secondary
MISU Receiver and Transmitter Capacity = 13 Sub-bands
Home ISU Receiverand Transmitter Capacity = 1 Sub-bands
Synchronization Channel
5 payload channels + 1 IOC channel + 5 payload channels
(24 Sub-bands)

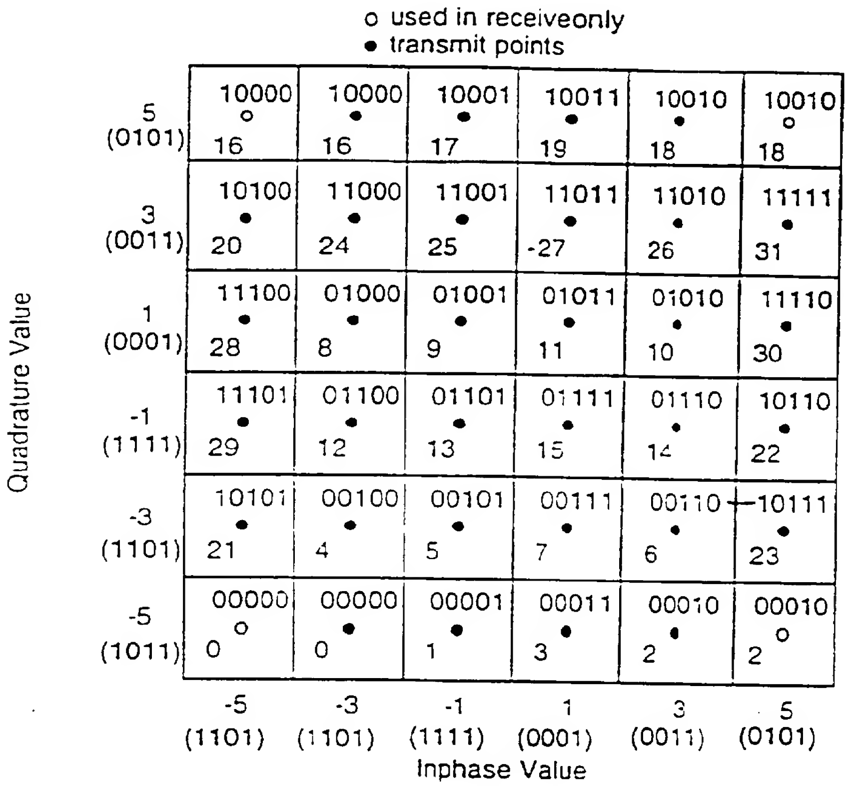


Figure 9B

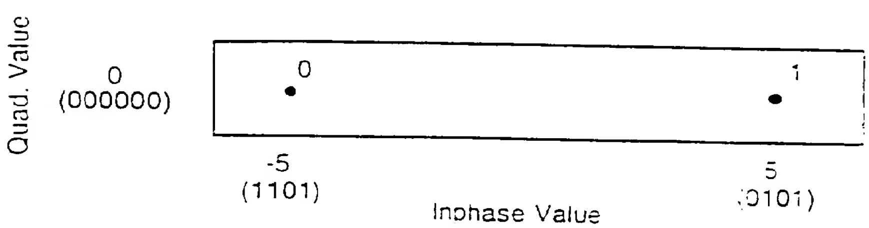


Figure 9C

Figure 10

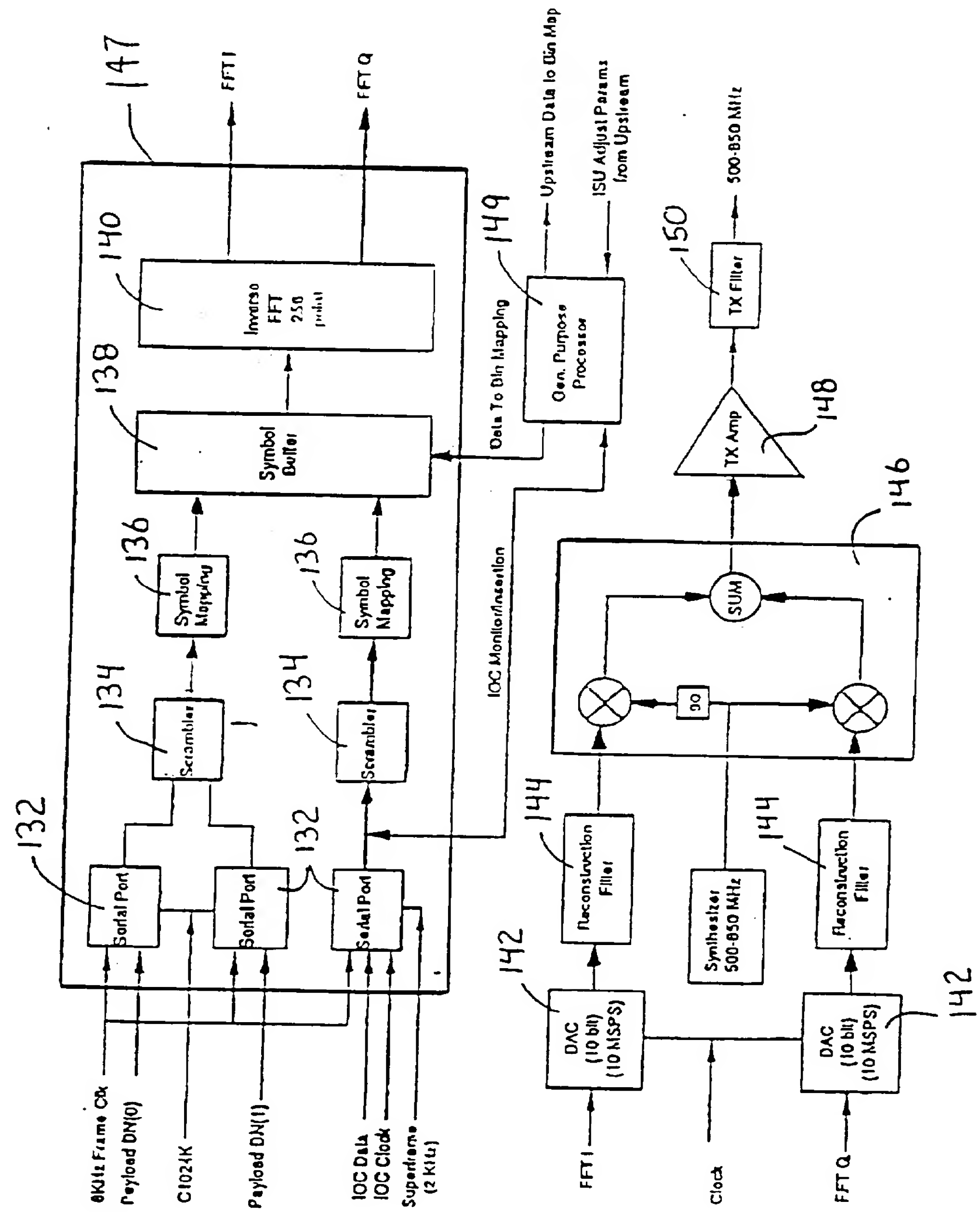

147
FFT I
FFT Q
140
Inverse FFT 256 point
138
Symbol Buffer
136
Symbol Mapping
134
Scrambler
132
Serial Port
149
Gen. Purpose Processor
Data To Bin Mapping
Upstream Data to Bin Map
ISU Adjust Params from Upstream
IOC Monitor/Insertion
150
TX Filter
500-850 MHz
148
TX Amp
146
SUM
90
144
Reconstruction Filter
Synthesizer 500-850 MHz
142
DAC (10 bit) (10 MSPS)
8KHz Frame Ck
Payload DN(0)
C1024K
Payload DN(1)
IOC Data
IOC Clock
Superframe (2 KHz)
FFT I
Clock
FFT Q

Figure 11

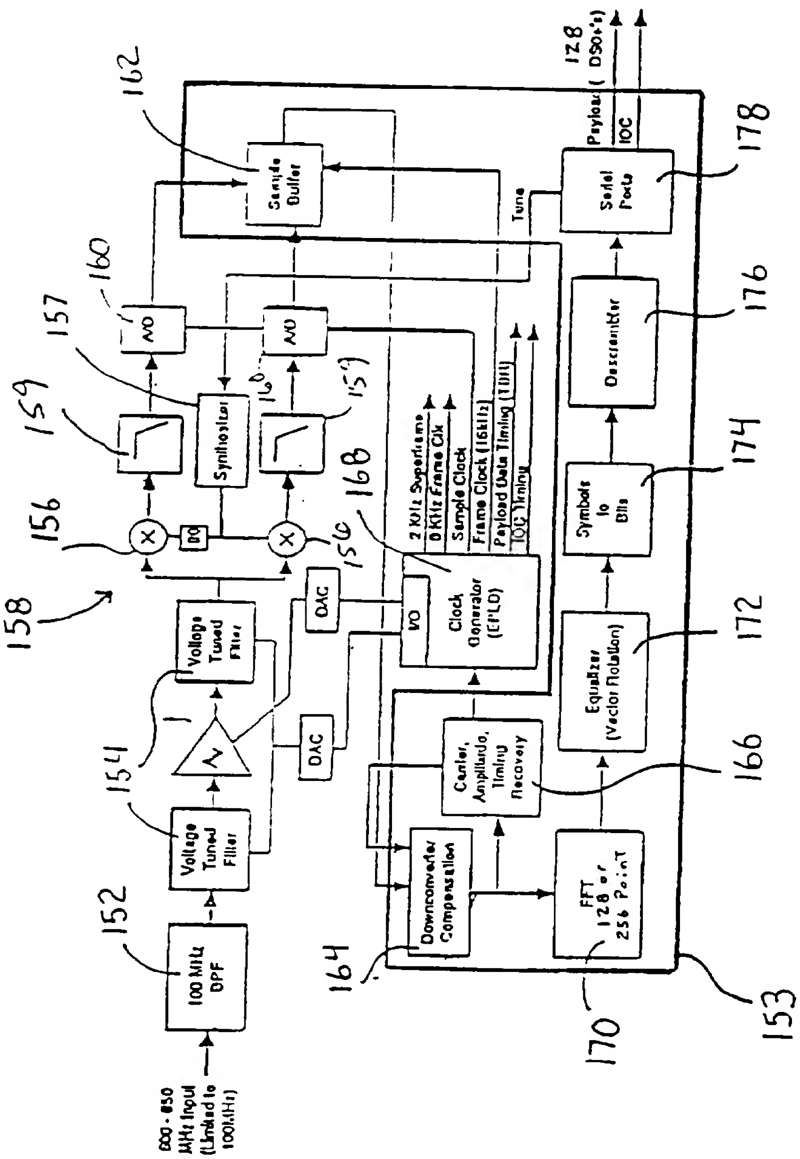
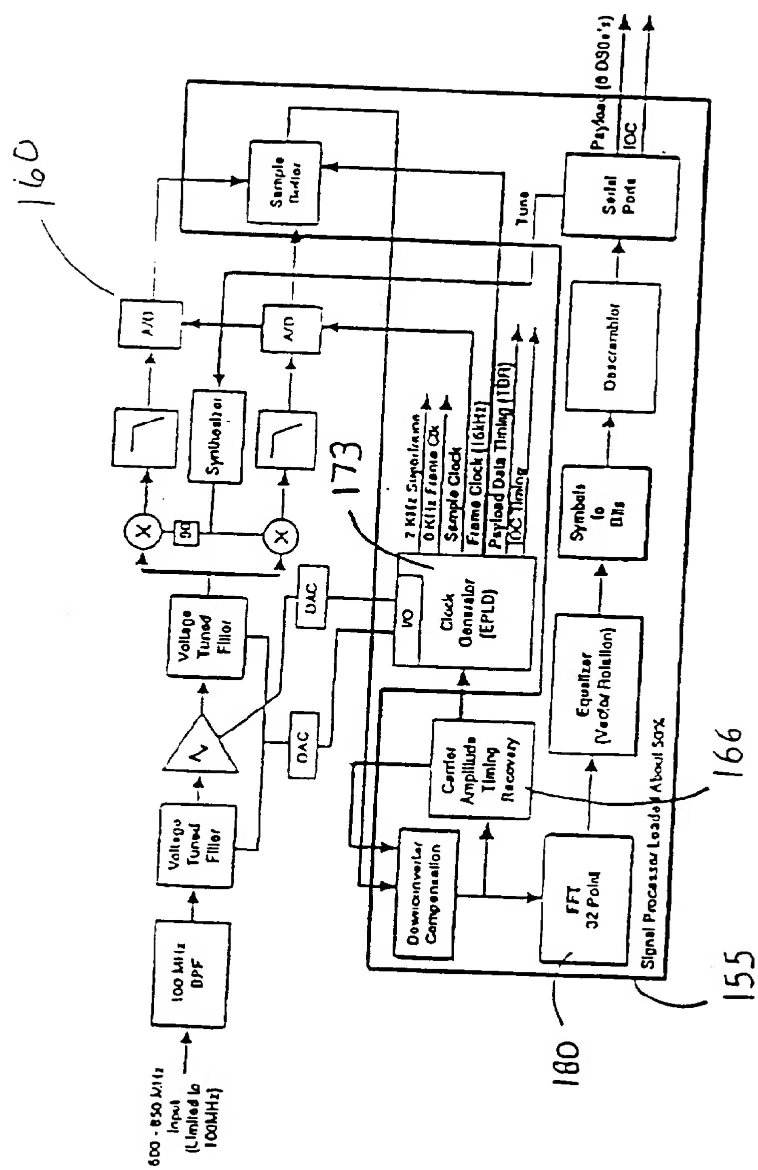

800-850 MHz Input (Limited to 100MHz)
100 MHz BPF
Voltage Tuned Filter
Voltage Tuned Filter
Synthesizer
A/D
A/D
Sample Buffer
DAC
DAC
VCO
Clock Generator (EPLD)
2 KHz Superframe
8 KHz Frame Clk
Sample Clock
Frame Clock (16kHz)
Payload Data Timing (TDI)
IOC Timing
Tune
Downconverter Compensation
Carrier, Amplitude, Timing Recovery
FFT 128 or 256 Point
Equalizer (Vector Rotation)
Symbols to Bits
Descrambler
Serial Ports
Payload
IOC
128 (DS0+'s)
152
153
154
156
156
157
158
159
159
160
160
162
164
166
168
170
172
174
176
178

Figure 12


100 MHz BPF
Voltage Tuned Filter
Voltage Tuned Filter
Synthesizer
A/D
A/D
Sample Buffer
DAC
DAC
I/O
Clock Generator (EPLD)
Sample Clock
Frame Clock (16kHz)
IOC Timing
Downconverter Compensation
FFT 32 Point
Carrier Amplitude Timing Recovery
Equalizer (Vector Rotation)
Symbols to Bits
Descrambler
Serial Ports
Tune
IOC
Signal Processor Loaded About 50%
160
173
166
155
180

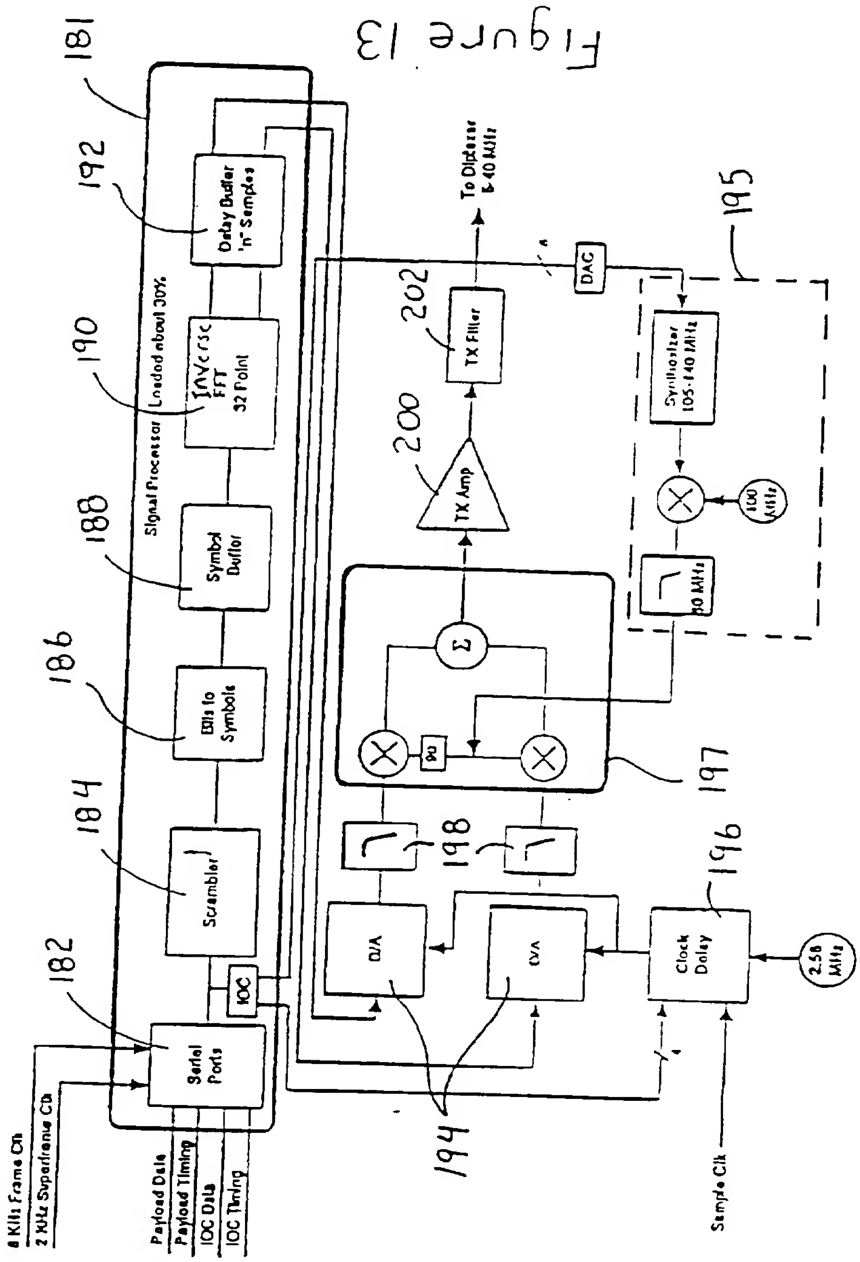

181
182
184
186
188
190
192
194
195
196
197
198
200
202
Signal Processor Loaded about 30%
Serial Ports
Scrambler
Bits to Symbols
Symbol Buffer
Inverse FFT 32 Point
Delay Buffer 'n' Samples
IOC
D/A
D/A
Clock Delay
2.58 MHz
Σ
90
TX Amp
TX Filter
To Diplexer 5-40 MHz
DAC
Synthesizer 105-140 MHz
100 MHz
30 MHz
8 KHz Frame Clk
2 KHz Superframe Clk
Payload Data
Payload Timing
IOC Data
IOC Timing
Sample Clk

Figure 13

Figure 14

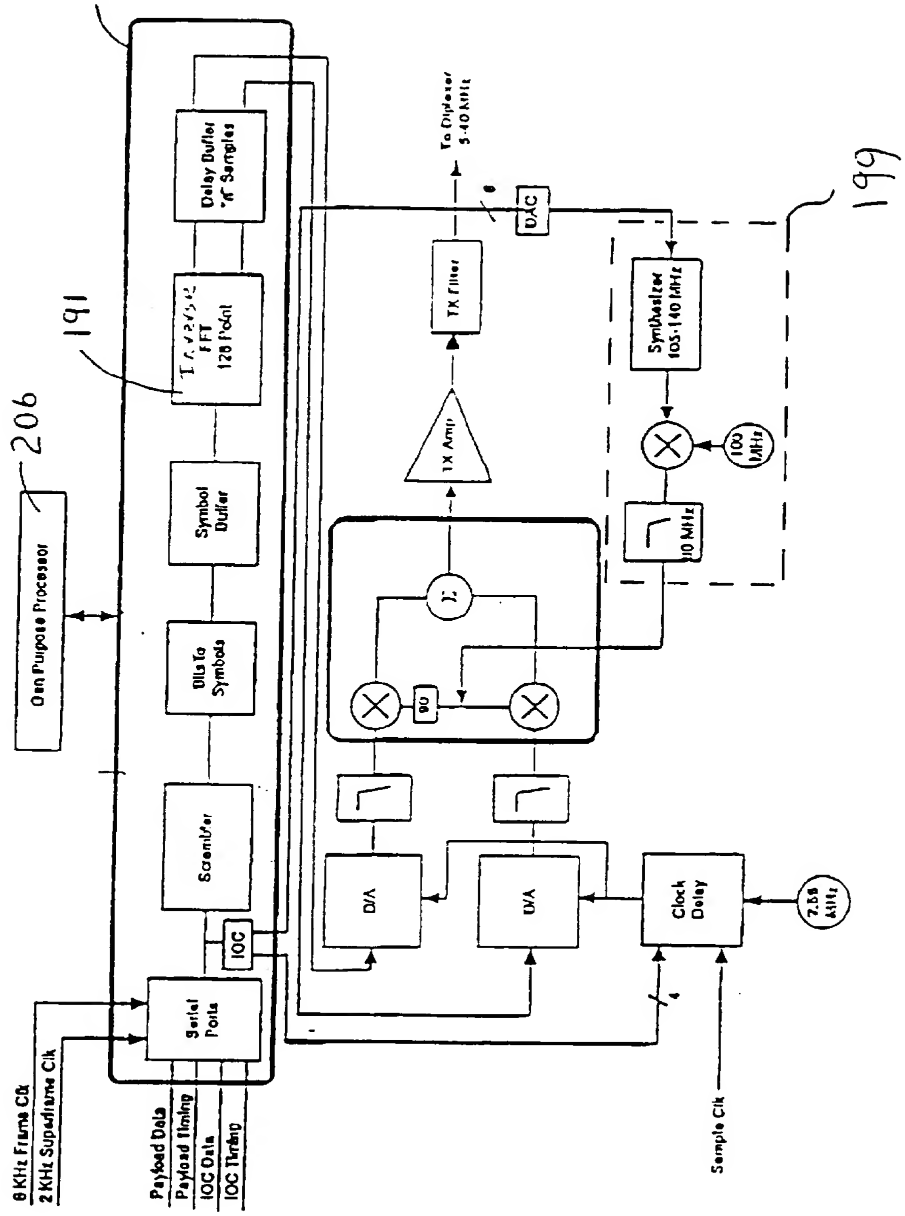
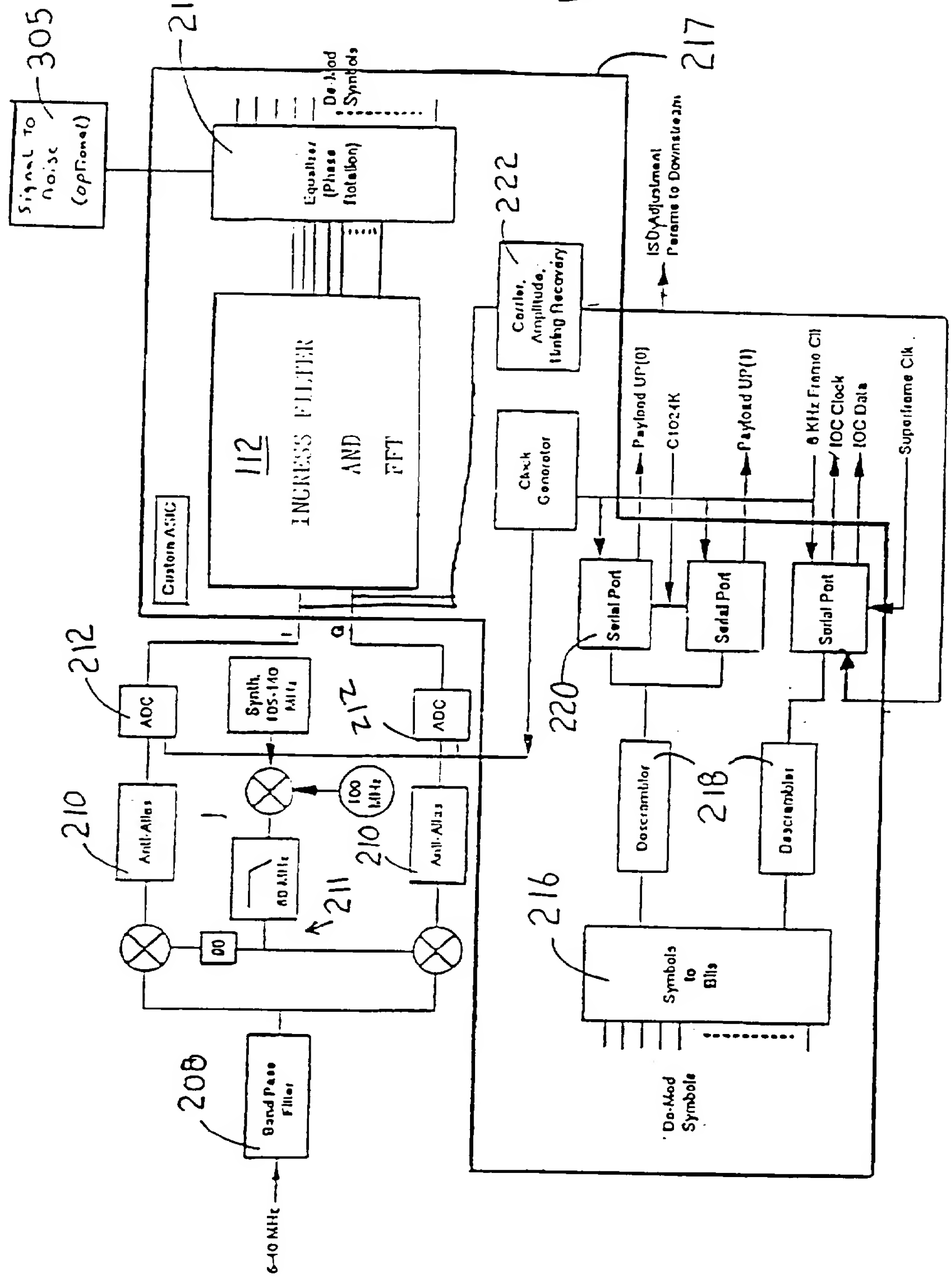

181
191
206
199
Gen Purpose Processor
Serial Ports
Scrambler
Bits To Symbols
Symbol Buffer
FFT
128 Point
IOC
D/A
Clock Delay
TX Amp
TX Filter
DAC
90
Synthesizer 105-140 MHz
100 MHz
10 MHz
7.88 MHz
Sample Clk
8 KHz Frame Clk
2 KHz Superframe Clk
Payload Data
Payload Timing
IOC Data
IOC Timing

Figure 15


Signal to Noise (optional)
305
214
Equalizer (Phase Rotation)
De-Mod Symbols
217
222
Carrier, Amplitude, Timing Recovery
ISDJ Adjustment Params to Downstream
112
INGRESS FILTER AND FFT
Custom ASIC
Clock Generator
Payload UP(0)
C1024K
Payload UP(1)
8 KHz Frame Clk
IOC Clock
IOC Data
Superframe Clk
Serial Port
Serial Port
Serial Port
220
I
Q
212
ADC
Synth. 105-140 MHz
212
ADC
210
Anti-Alias
100 MHz
40 MHz
90
211
210
Anti-Alias
218
Descrambler
Descrambler
216
Symbols to Bits
De-Mod Symbols
208
Band Pass Filter
6-40 MHz

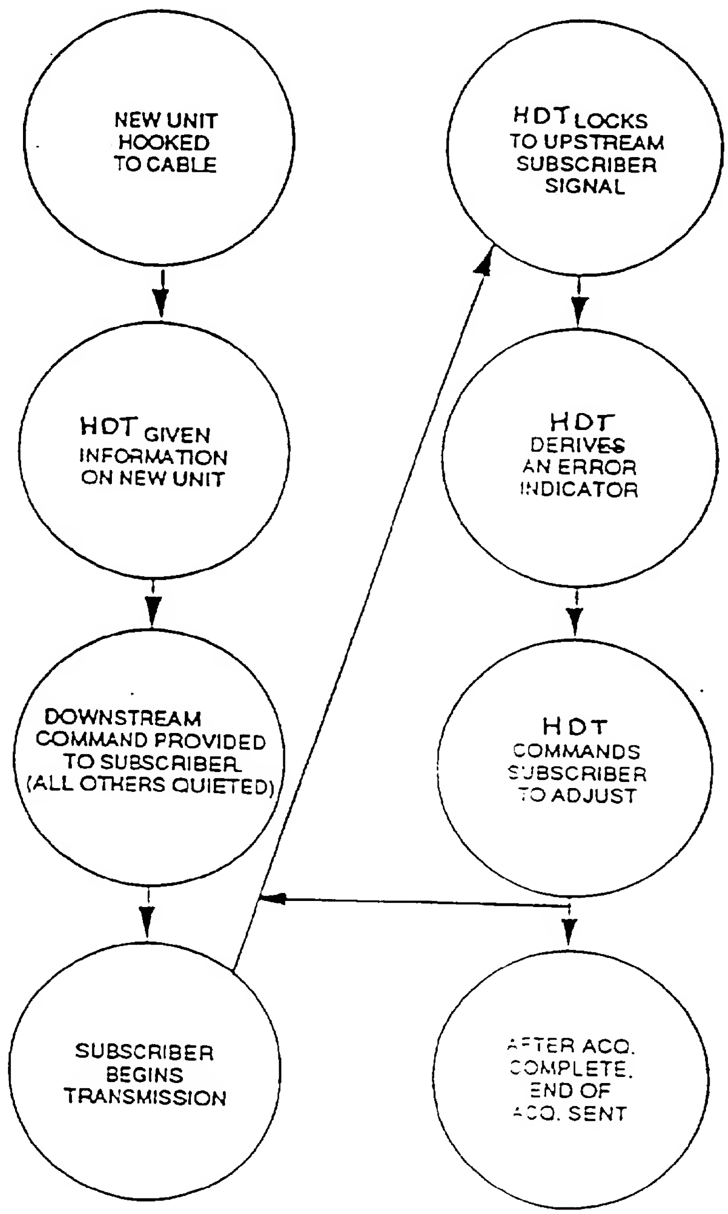

NEW UNIT HOOKED TO CABLE
HDT GIVEN INFORMATION ON NEW UNIT
DOWNSTREAM COMMAND PROVIDED TO SUBSCRIBER (ALL OTHERS QUIETED)
SUBSCRIBER BEGINS TRANSMISSION
HDT LOCKS TO UPSTREAM SUBSCRIBER SIGNAL
HDT DERIVES AN ERROR INDICATOR
HDT COMMANDS SUBSCRIBER TO ADJUST
AFTER ACQ. COMPLETE. END OF ACQ. SENT

Figure 16

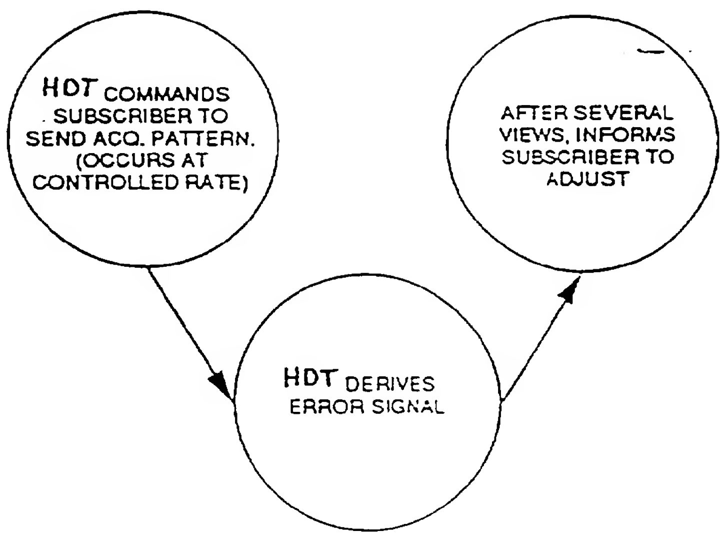
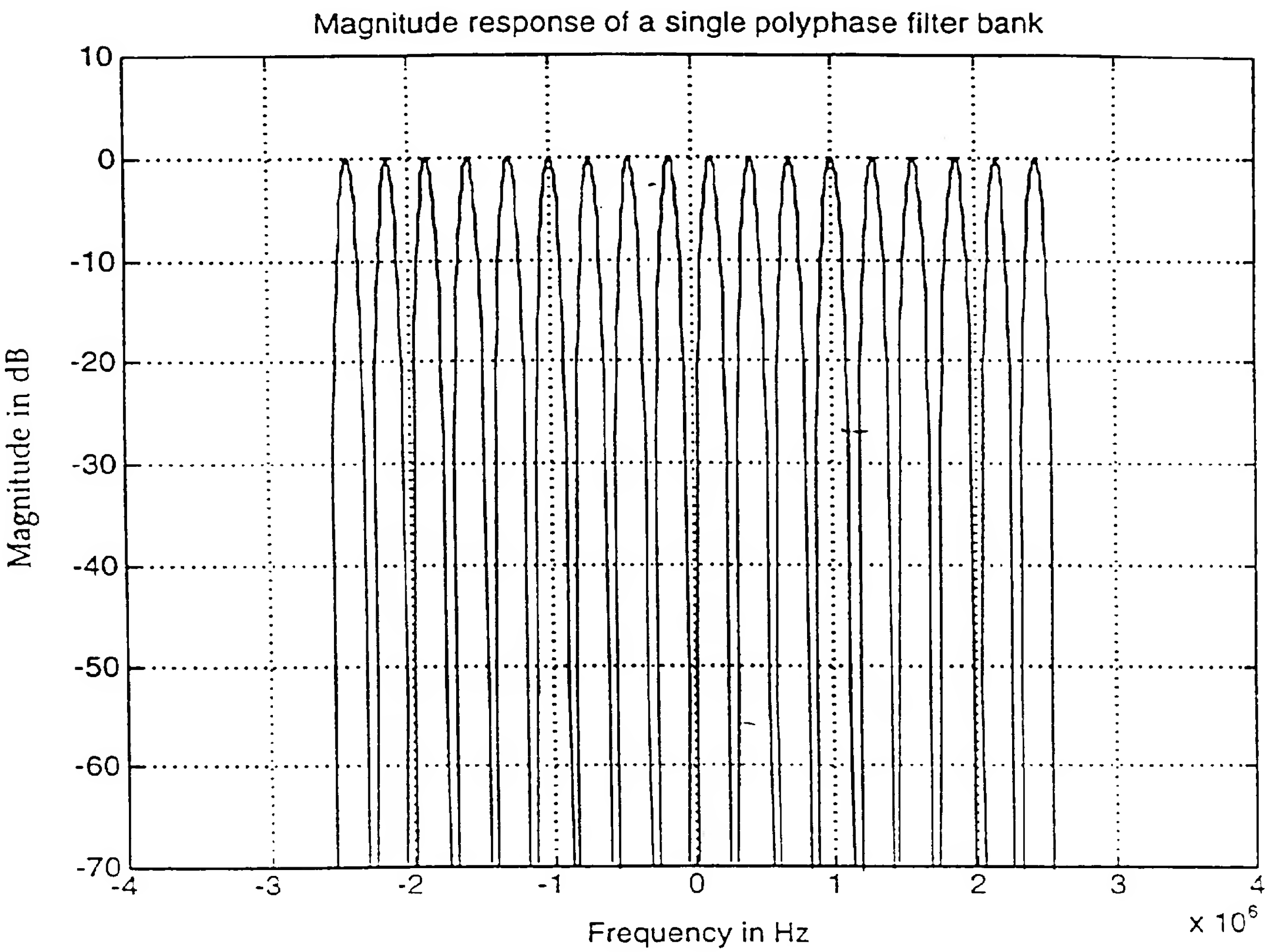

HDT COMMANDS SUBSCRIBER TO SEND ACQ. PATTERN. (OCCURS AT CONTROLLED RATE)
HDT DERIVES ERROR SIGNAL
AFTER SEVERAL VIEWS, INFORMS SUBSCRIBER TO ADJUST

Figure 17



Magnitude response of a single polyphase filter bank
Magnitude in dB
10
0
-10
-20
-30
-40
-50
-60
-70
-4
-3
-2
-1
0
1
2
3
4
x 10^6
Frequency in Hz

Figure 18

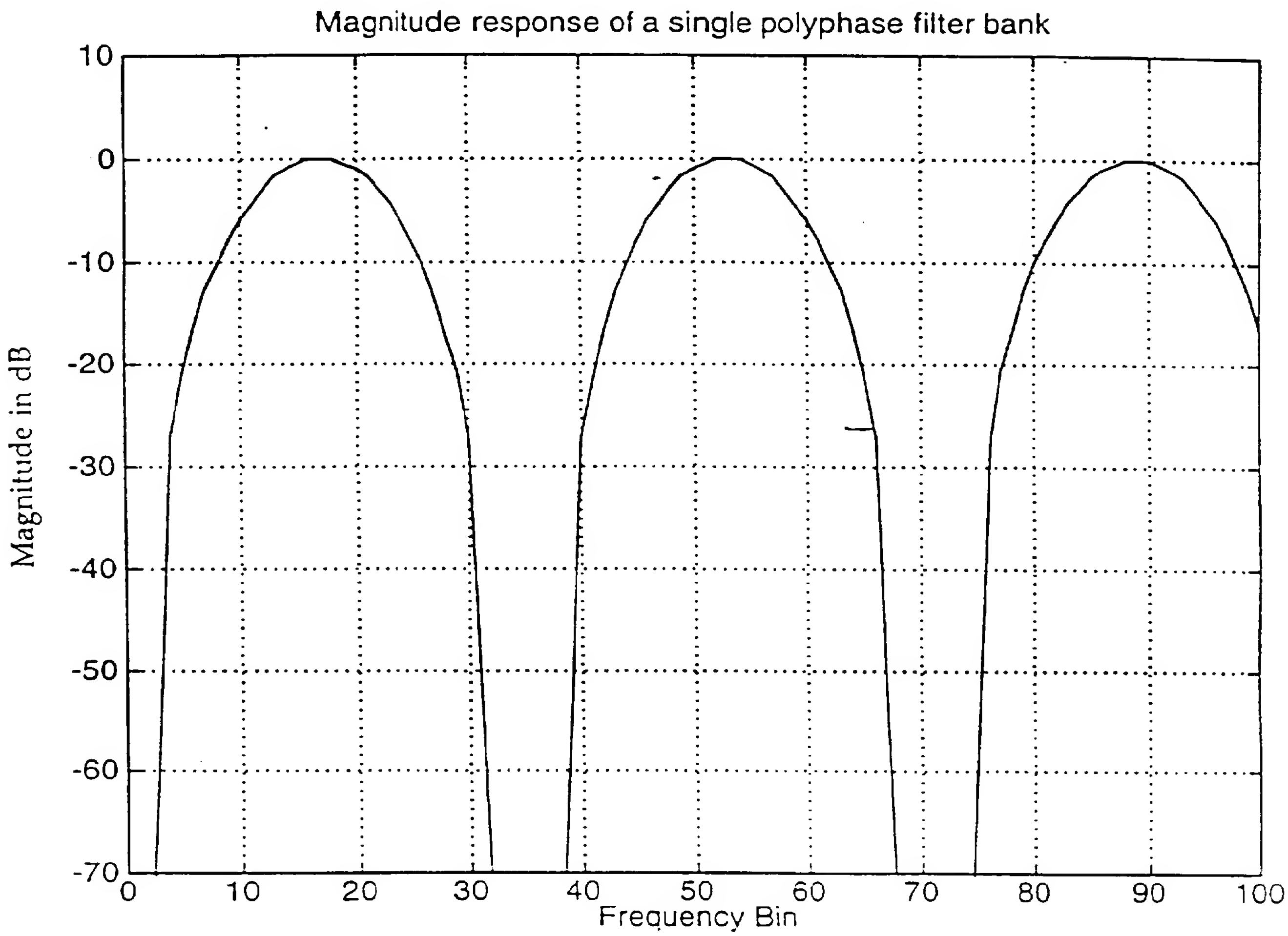
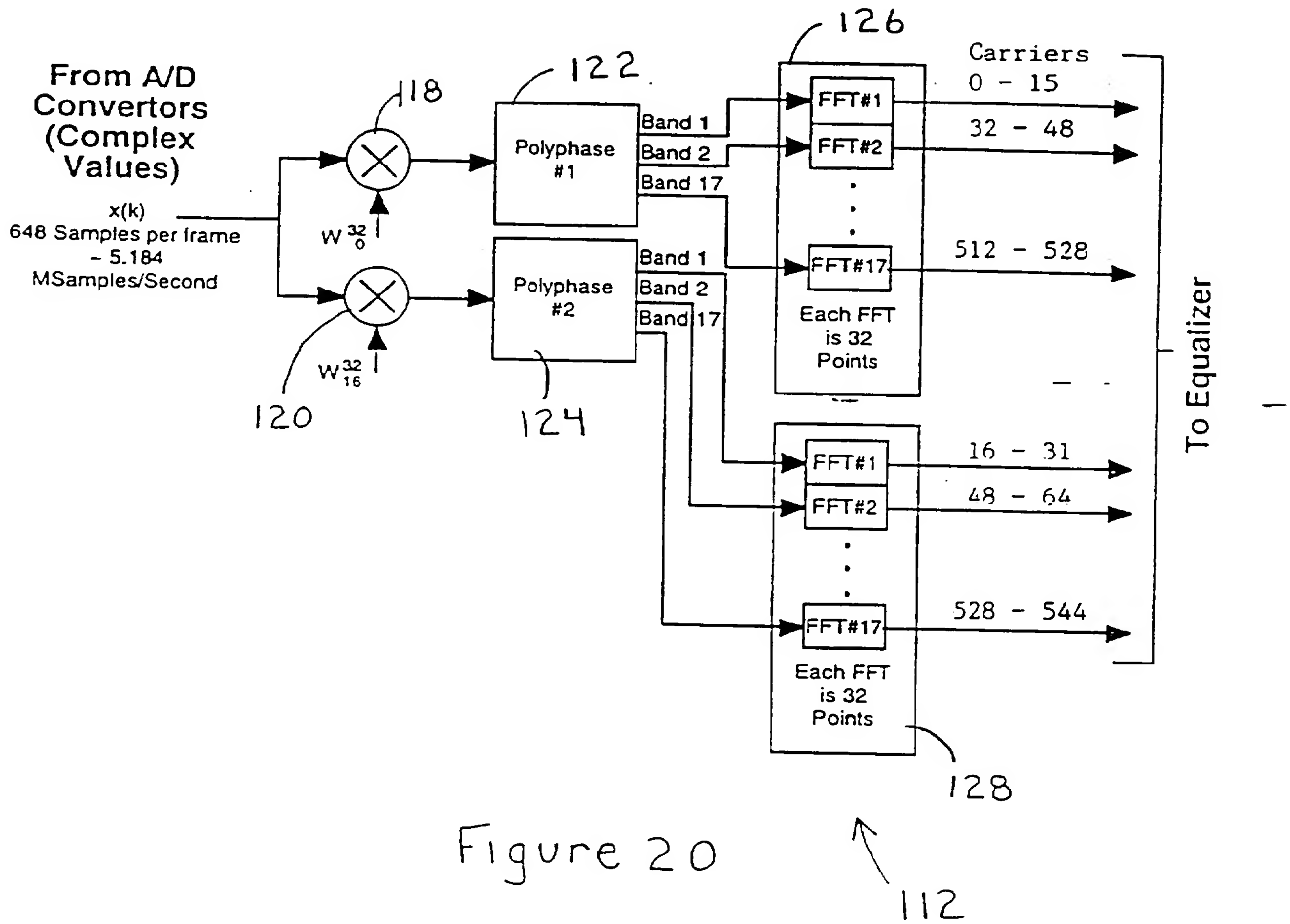
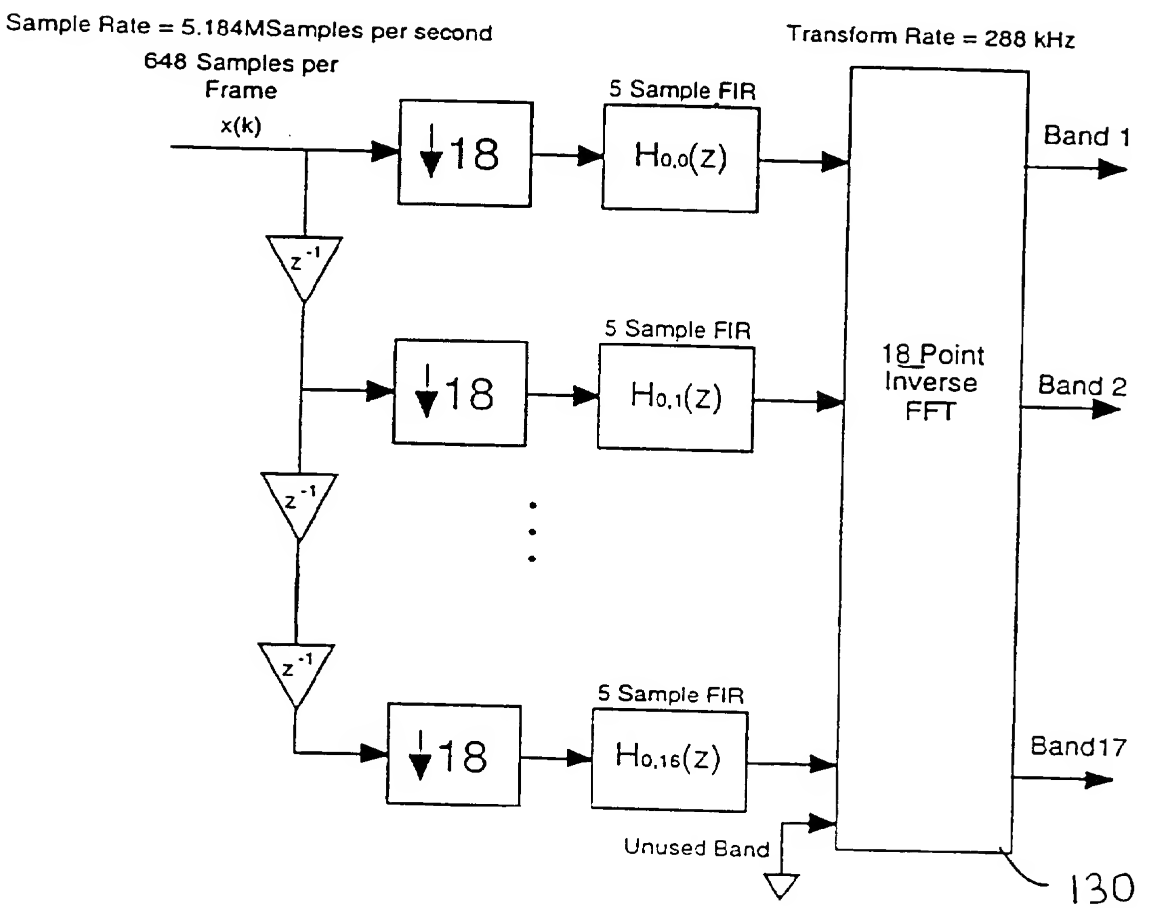

Magnitude response of a single polyphase filter bank
10
0
-10
-20
-30
-40
-50
-60
-70
Magnitude in dB
0
10
20
30
40
50
60
70
80
90
100
Frequency Bin

Figure 19



From A/D Convertors (Complex Values)
x(k)
648 Samples per frame
- 5.184
MSamples/Second
118
120
W 32 0
W 32 16
122
124
Polyphase #1
Polyphase #2
Band 1
Band 2
Band 17
126
128
FFT#1
FFT#2
FFT#17
Each FFT is 32 Points
Carriers
0 - 15
32 - 48
512 - 528
16 - 31
48 - 64
528 - 544
To Equalizer
112

Figure 20



Sample Rate = 5.184MSamples per second
648 Samples per Frame
x(k)
Transform Rate = 288 kHz
5 Sample FIR
↓18
$H_{0,0}(z)$
z^{-1}
5 Sample FIR
↓18
$H_{0,1}(z)$
z^{-1}
z^{-1}
5 Sample FIR
↓18
$H_{0,16}(z)$
Unused Band
18 Point Inverse FFT
Band 1
Band 2
Band17
130
122, 124

Figure 21

Figure 22 A

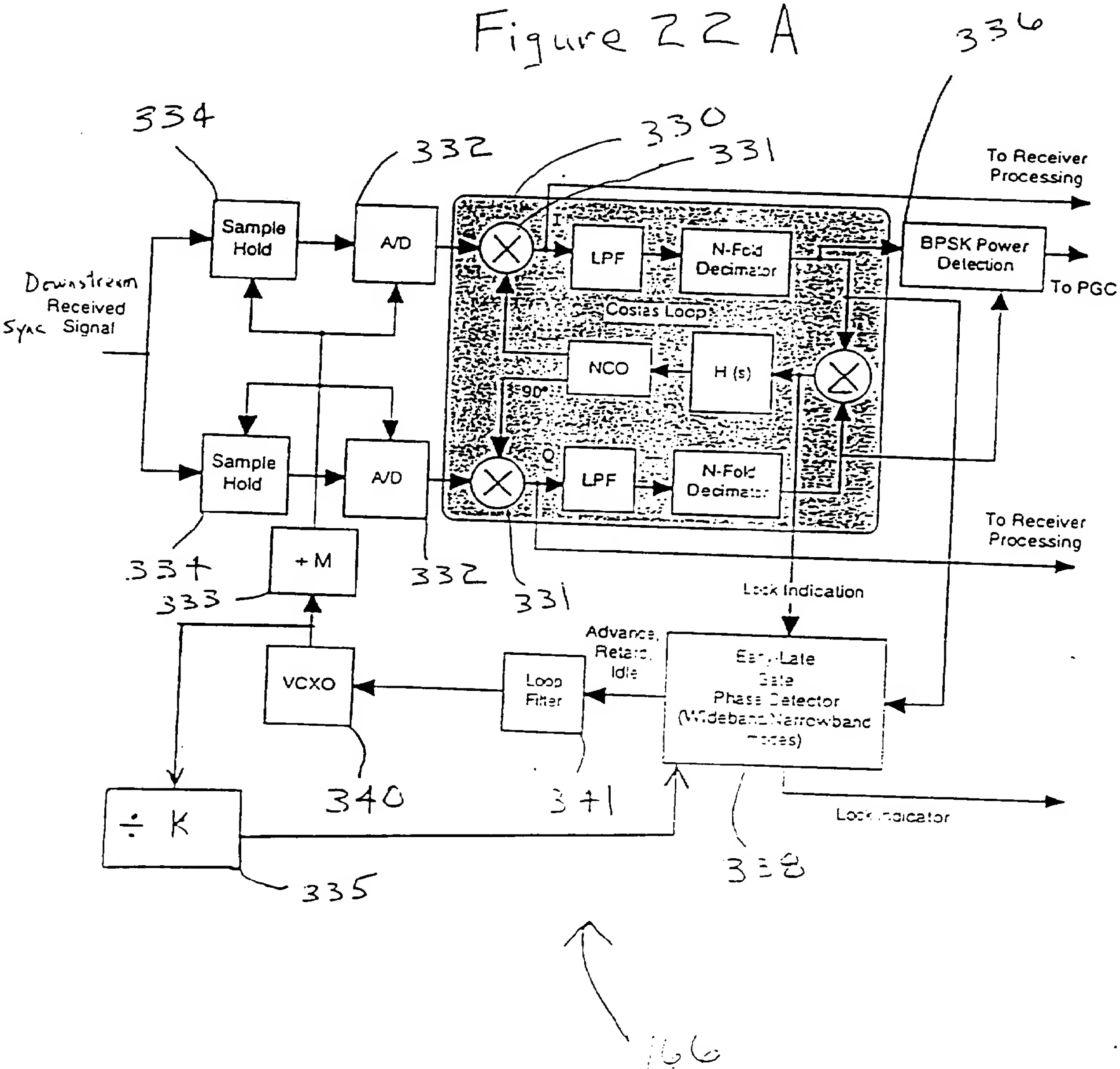

336
334
332
330
331
To Receiver Processing
Sample Hold
A/D
LPF
N-Fold Decimator
BPSK Power Detection
To PGC
Downstream Received Sync Signal
Costas Loop
NCO
H (s)
90°
Q
Sample Hold
A/D
LPF
N-Fold Decimator
To Receiver Processing
334
333
+M
332
331
Lock Indication
Advance, Retard, Idle
VCXO
Loop Filter
Early-Late Gate Phase Detector (Wideband/Narrowband modes)
340
341
÷ K
Lock Indicator
338
335
166

Figure 22B

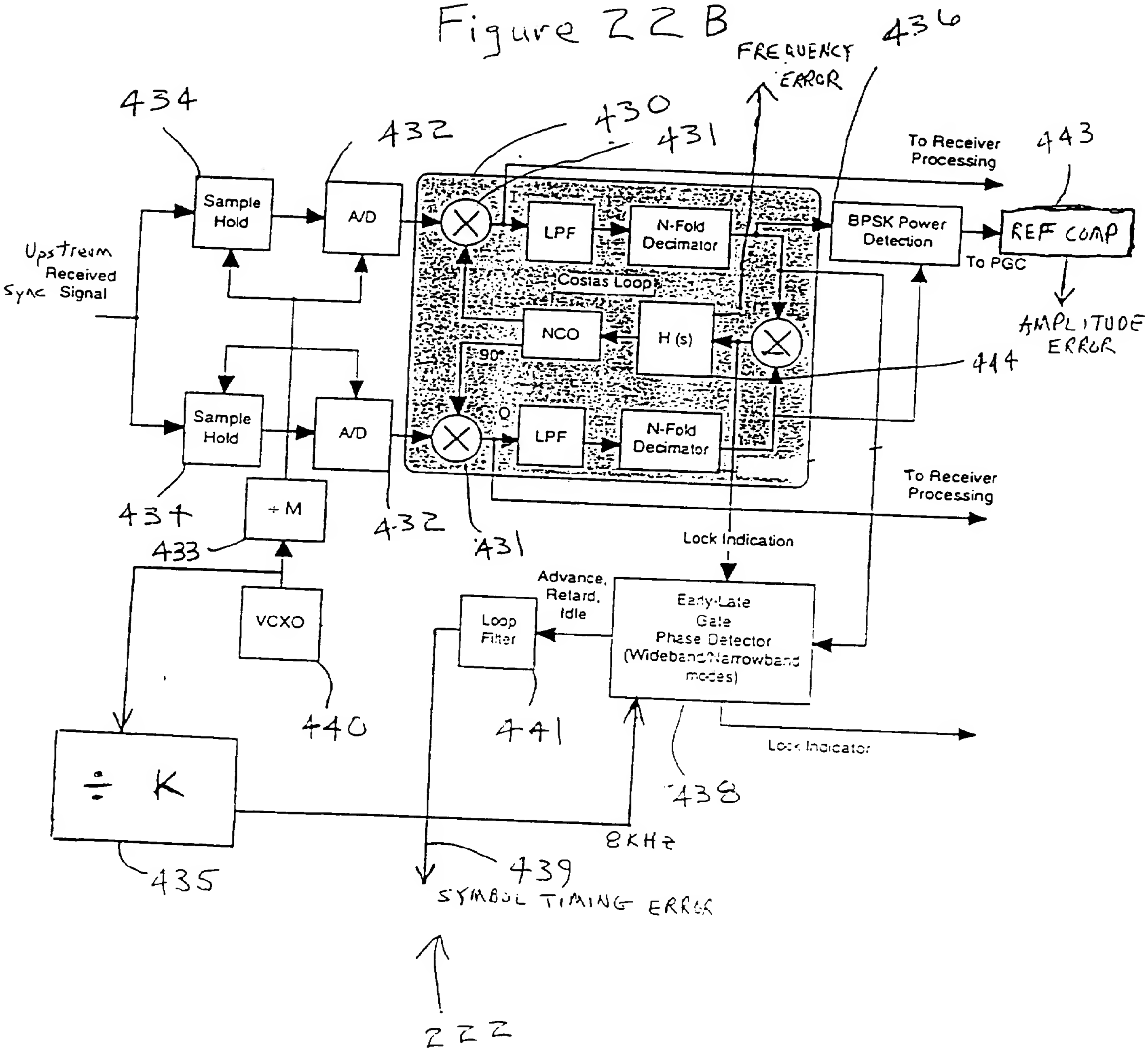
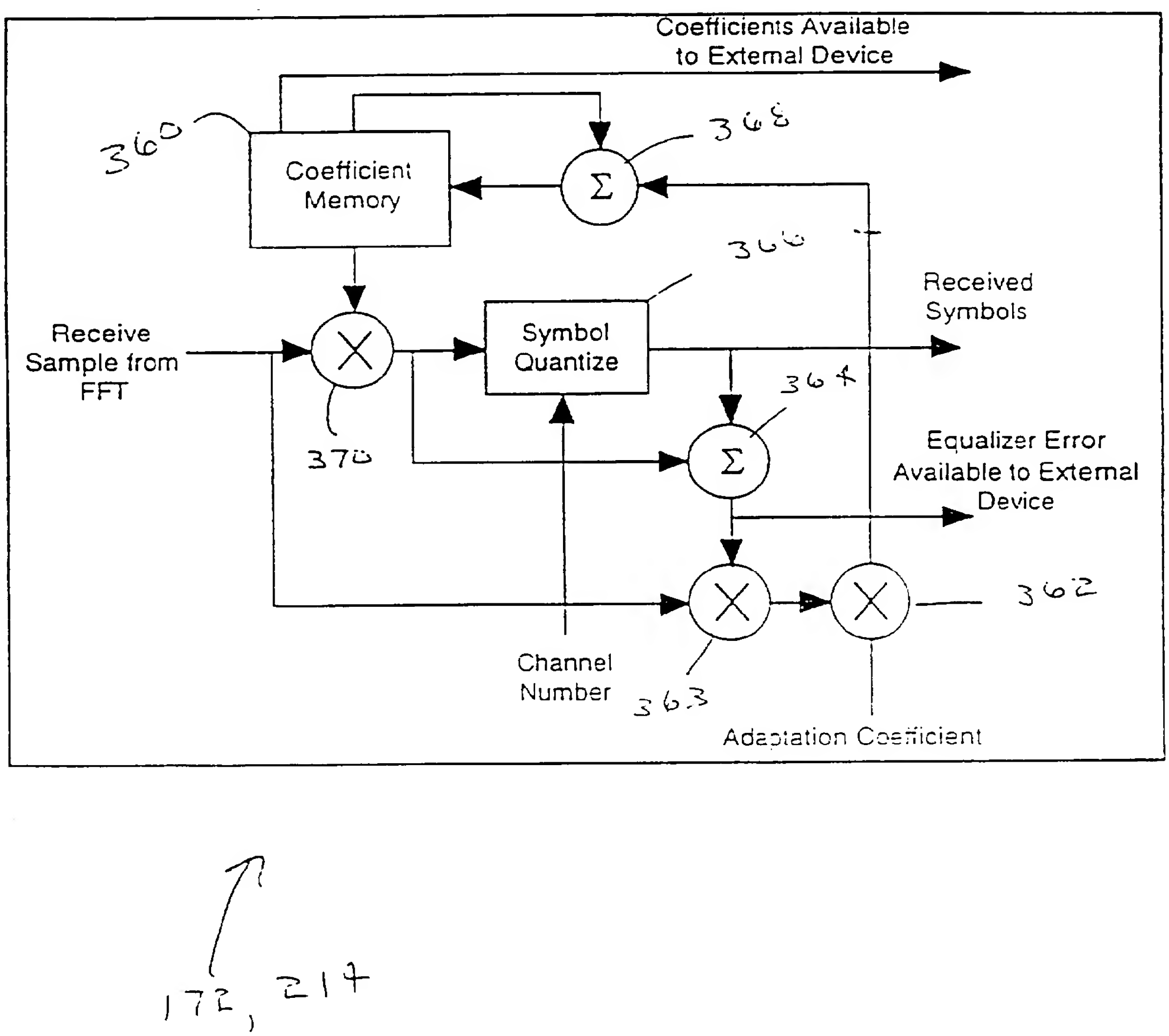

FREQUENCY ERROR
436
434
432
430
431
To Receiver Processing
443
Upstream Received Sync Signal
Sample Hold
A/D
LPF
N-Fold Decimator
BPSK Power Detection
REF COMP
To PGC
Costas Loop
NCO
H (s)
AMPLITUDE ERROR
90°
444
Q
Sample Hold
A/D
LPF
N-Fold Decimator
To Receiver Processing
437
÷ M
433
432
431
Lock Indication
Advance, Retard, Idle
Loop Filter
Early-Late Gate Phase Detector (Wideband/Narrowband modes)
VCXO
440
441
Lock Indicator
÷ K
438
8KHz
435
439
SYMBOL TIMING ERROR
222

Figure 23


Coefficients Available to External Device
360
368
Coefficient Memory
Σ
366
Received Symbols
Receive Sample from FFT
Symbol Quantize
364
370
Σ
Equalizer Error Available to External Device
362
Channel Number
363
Adaptation Coefficient
172, 217

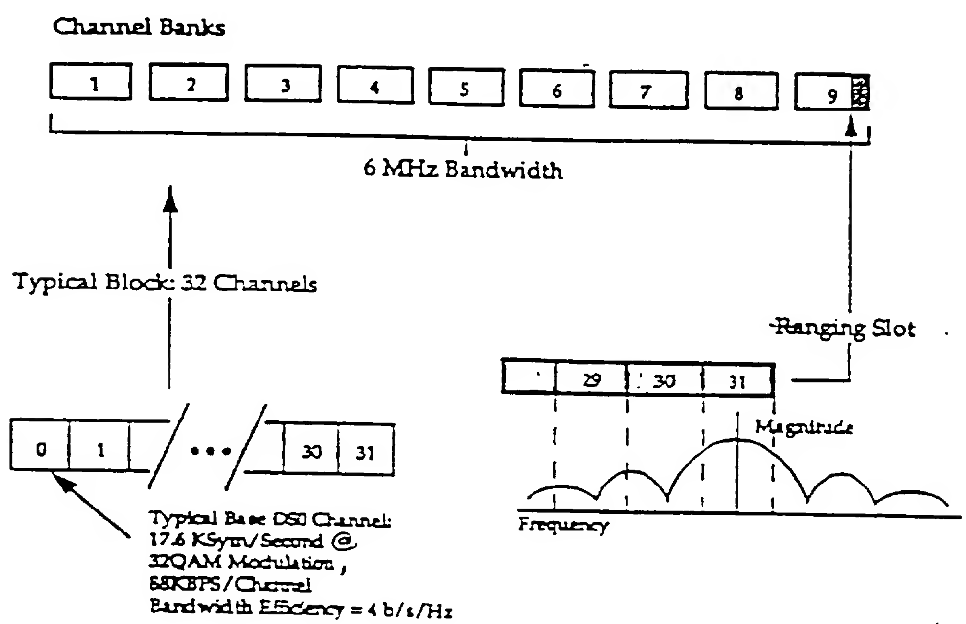

Channel Banks
1
2
3
4
5
6
7
8
9
6 MHz Bandwidth
Typical Block: 32 Channels
Ranging Slot
29
30
31
Magnitude
0
1
30
31
Frequency
Typical Base DS0 Channel:
17.6 KSym/Second @
32QAM Modulation,
68KBPS/Channel
Bandwidth Efficiency = 4 b/s/Hz

Figure 24

Figure 25

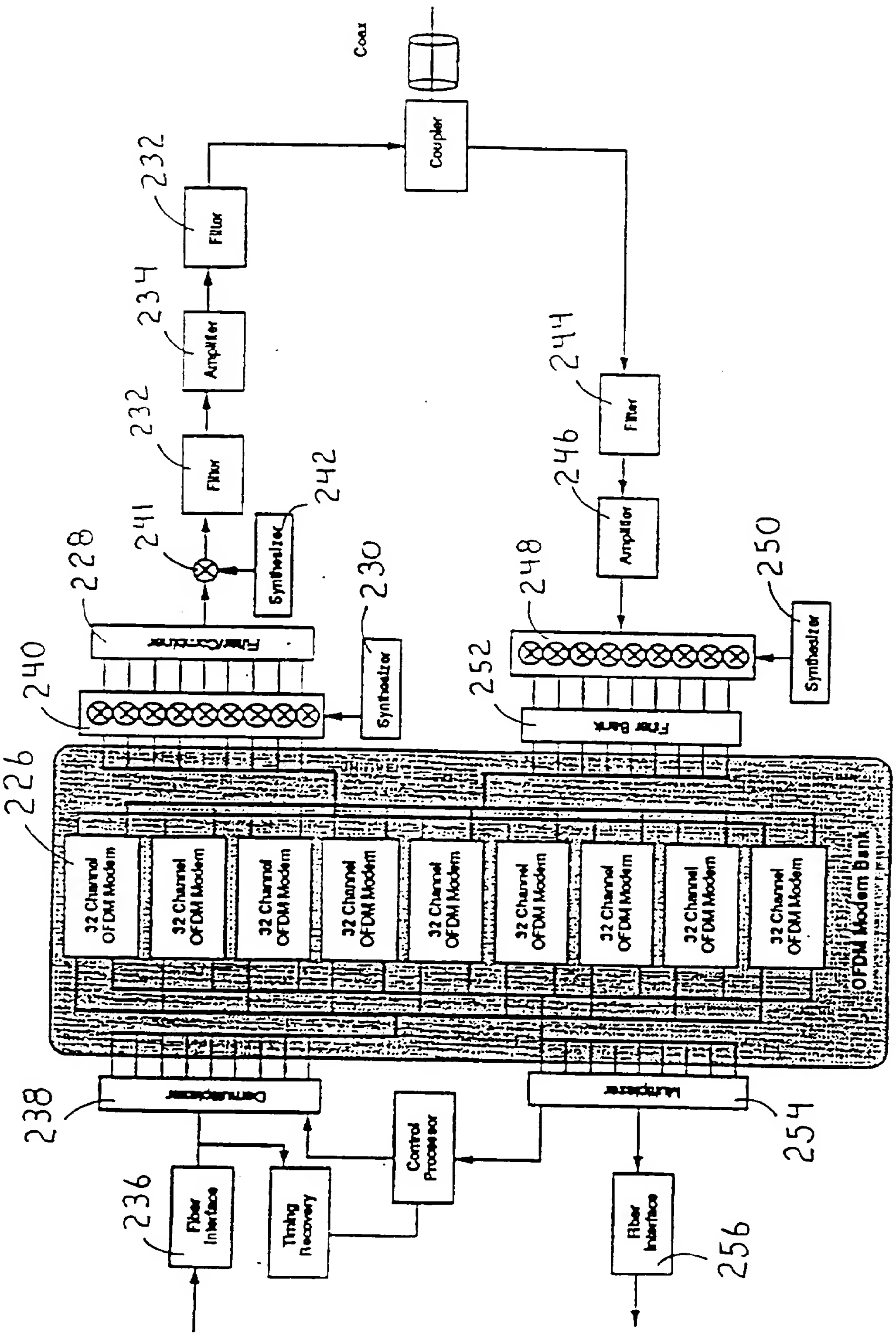
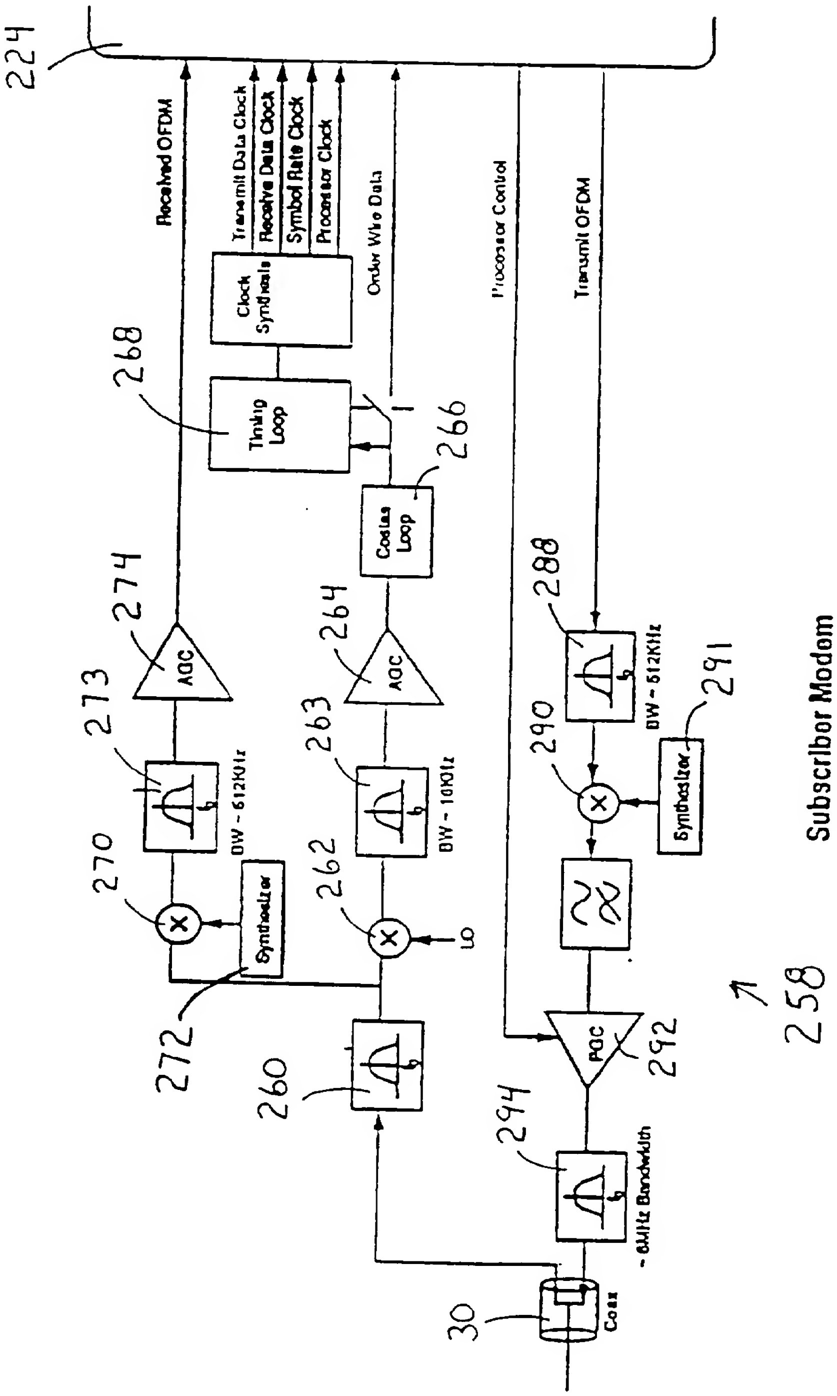

Coax
Coupler
Filter
Amplifier
Filter
Synthesizer
Filter/Combiner
Synthesizer
Filter
Amplifier
Synthesizer
Filter Bank
32 Channel OFDM Modem
OFDM Modem Bank
Demultiplexer
Multiplexer
Control Processor
Fiber Interface
Timing Recovery
Fiber Interface
232
234
232
241
242
228
230
240
226
244
246
248
250
252
238
236
254
256

Figure 26


224
Received OFDM
Transmit Data Clock
Receive Data Clock
Symbol Rate Clock
Processor Clock
Order Wire Data
Processor Control
Transmit OFDM
Clock Synthesis
268
Timing Loop
266
Costas Loop
274
AGC
264
AGC
288
BW - 512KHz
273
263
291
BW - 512KHz
BW - 10KHz
290
Synthesizer
270
262
Synthesizer
LO
272
260
292
PAC
294
~6MHz Bandwidth
30
Coax
Subscriber Modem
258

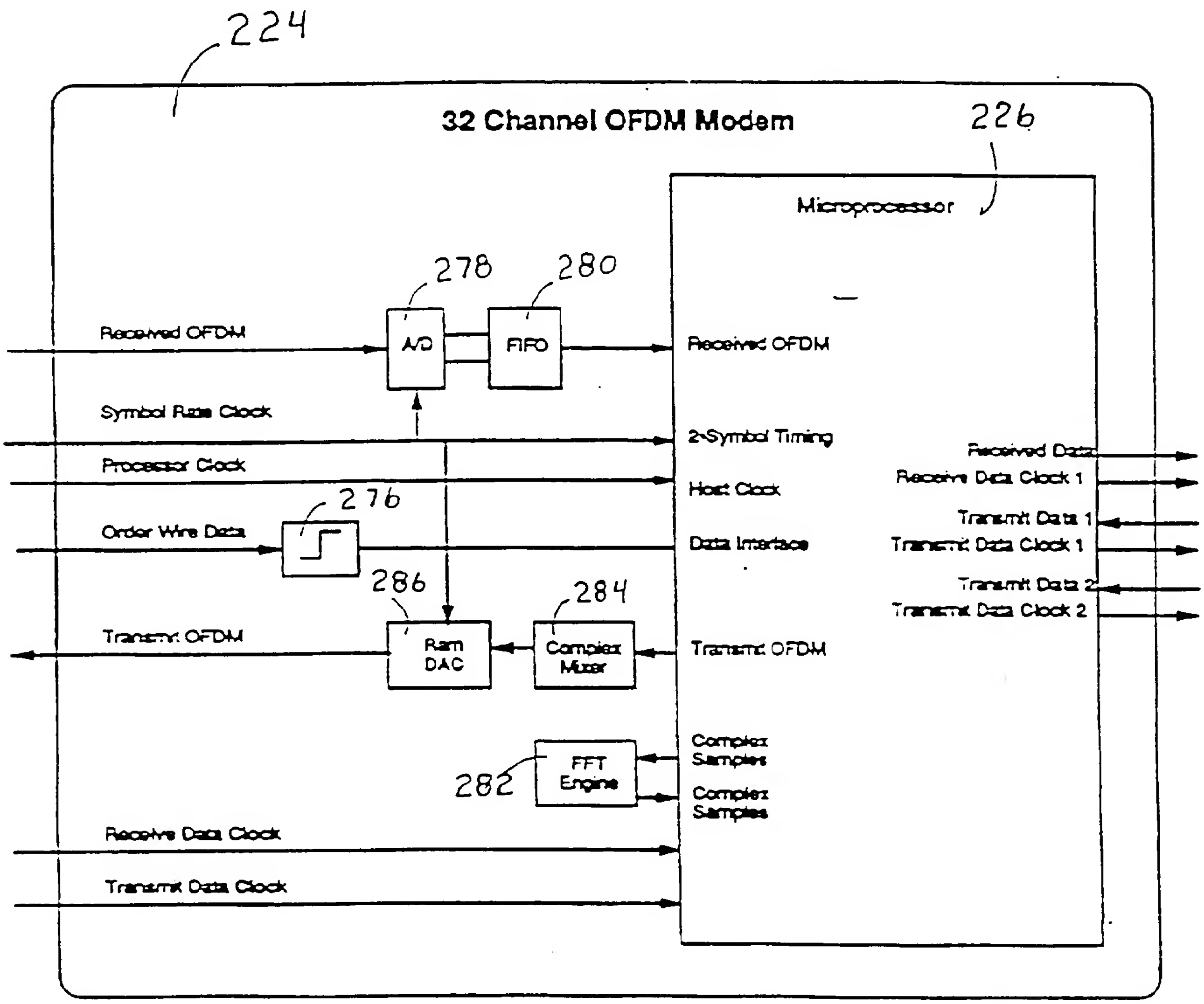

224
32 Channel OFDM Modem
226
Microprocessor
278
280
Received OFDM
A/D
FIFO
Received OFDM
Symbol Rate Clock
2xSymbol Timing
Processor Clock
Host Clock
276
Order Wire Data
Data Interface
286
284
Transmit OFDM
Ram DAC
Complex Mixer
Transmit OFDM
FFT Engine
282
Complex Samples
Complex Samples
Receive Data Clock
Transmit Data Clock
Received Data
Receive Data Clock 1
Transmit Data 1
Transmit Data Clock 1
Transmit Data 2
Transmit Data Clock 2

Figure 27

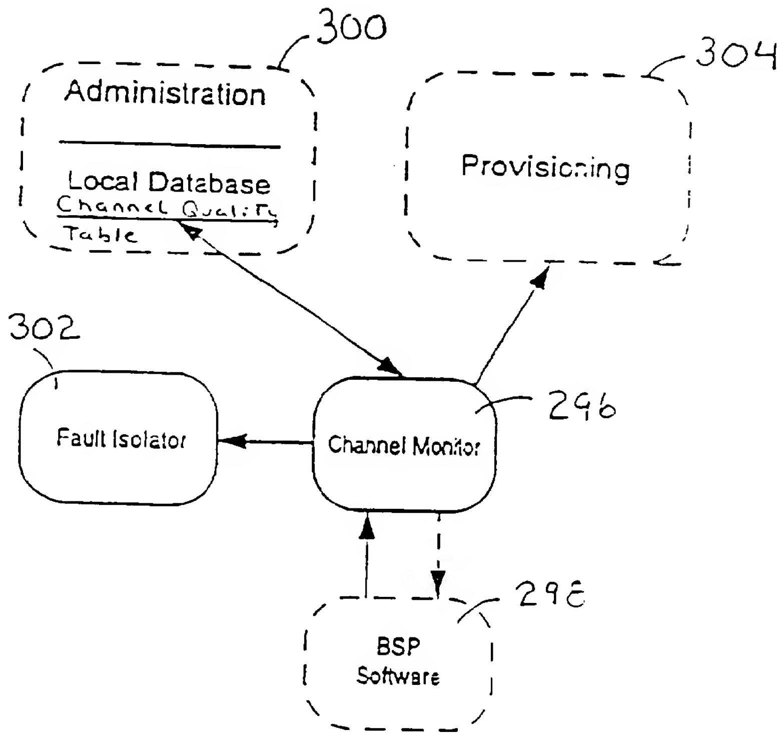

300
Administration
Local Database
Channel Quality Table
304
Provisioning
302
Fault Isolator
Channel Monitor
296
298
BSP Software

Figure 28

Figure 29A

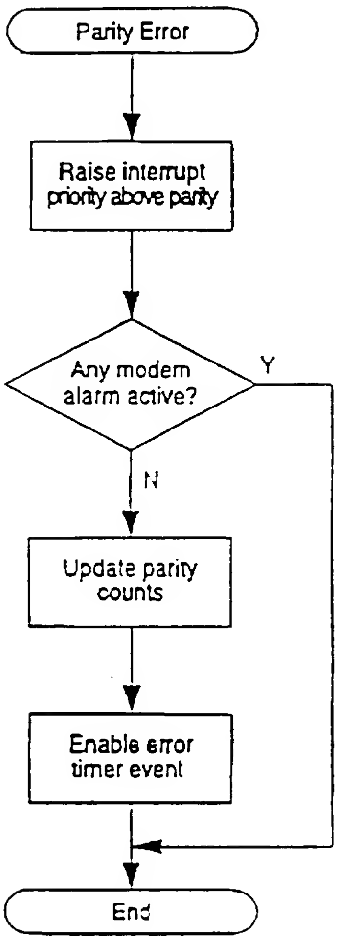
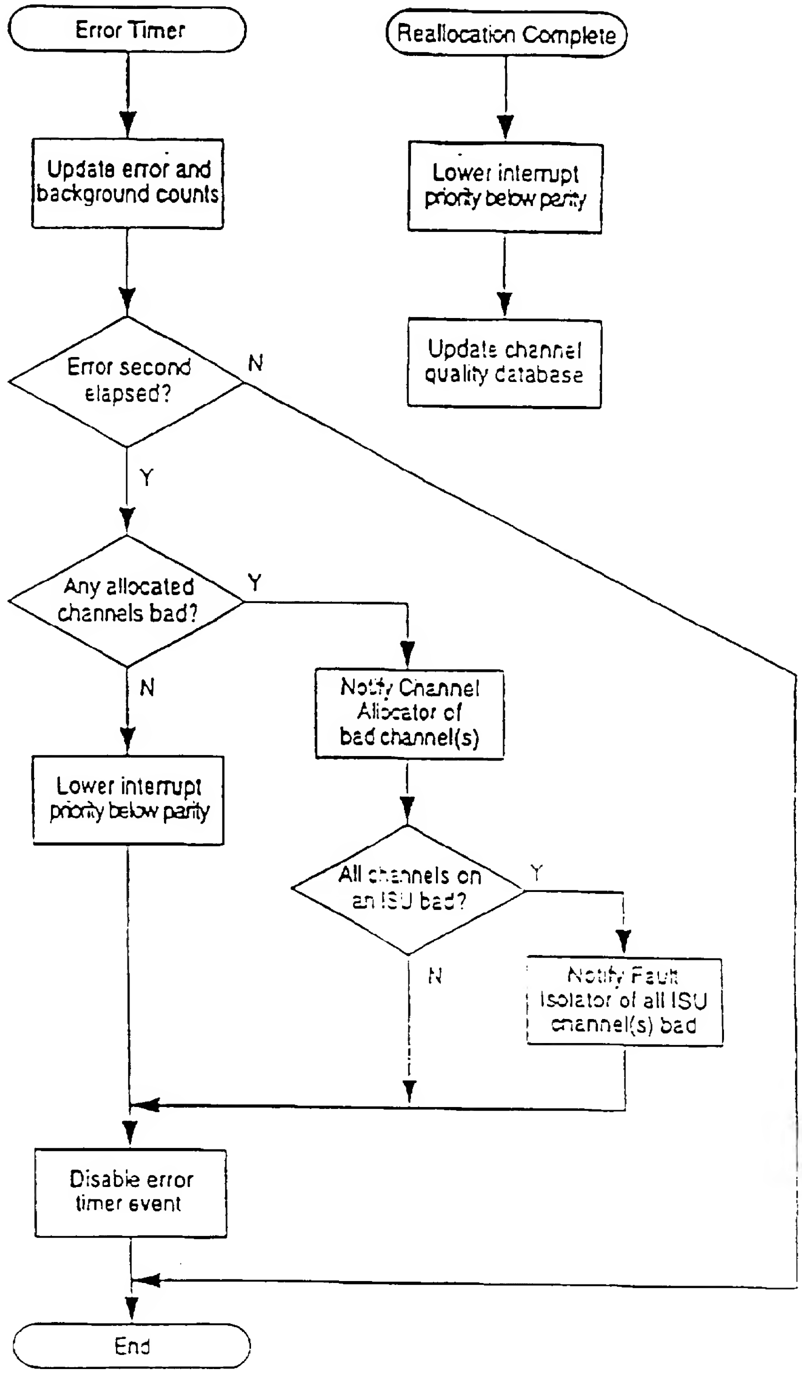


Figure 29B Figure 29C



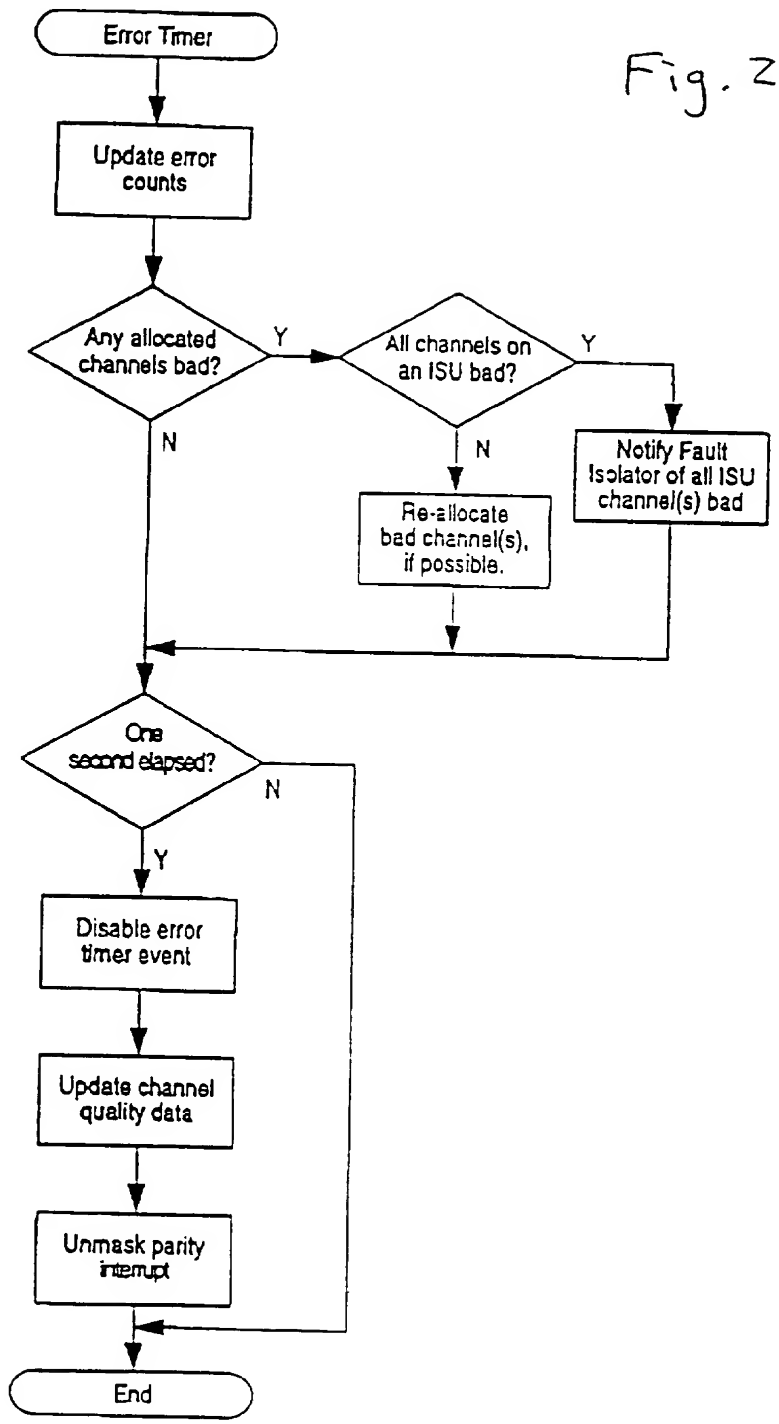

Error Timer
Update error counts
Any allocated channels bad?
Y
N
All channels on an ISU bad?
Y
N
Notify Fault Isolator of all ISU channel(s) bad
Re-allocate bad channel(s), if possible.
One second elapsed?
N
Y
Disable error timer event
Update channel quality data
Unmask parity interrupt
End

Fig. 29D

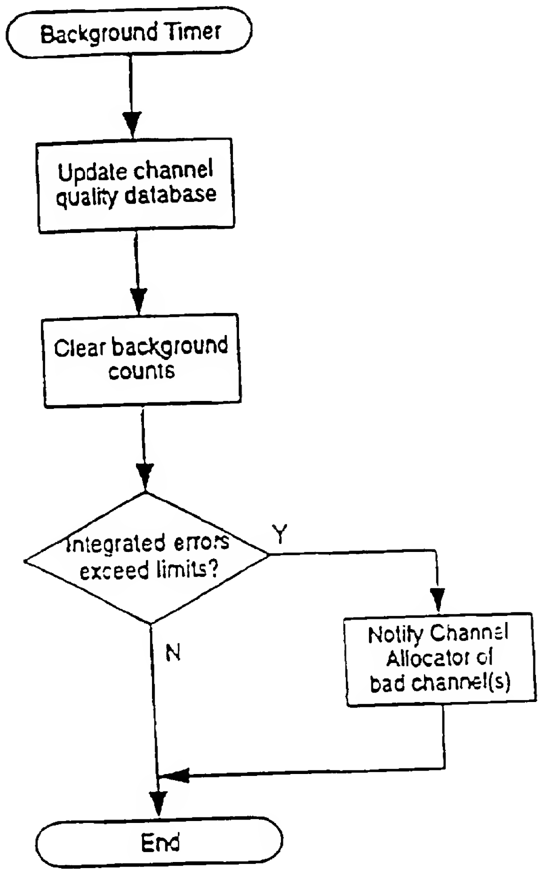


Figure 30

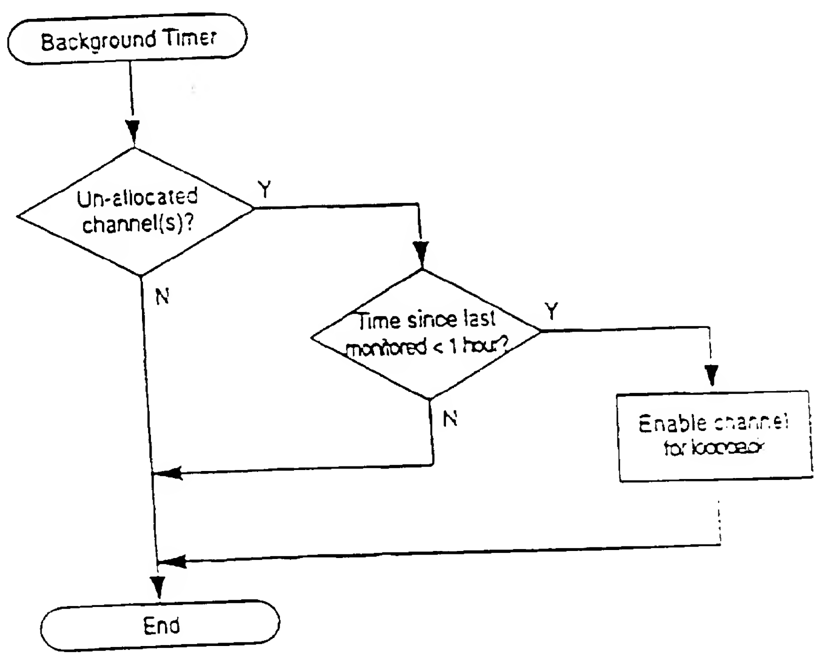


Figure 31